SVG401S 4 FXS SIP Phone Adapter User Manual

Version 1.1

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1 Welcome

The SVG401S is an Analog Phone Adapter (ATA) that integrates Networking and VoIP, which enables users experience VoIP communication while surfing the internet. It is fully compatible with SIP standard and can interoperate with other SIP compliant devices and software.

2 Installation

2.1 Package List

The SVG401S ATA package contains:

1) One SVG401S ATA

2) One universal power adapter

3) One Straight Ethernet cable

2.2 Safety Compliances

The SVG401S Sip ATA should be operated only with the universal power adapter provided with the package. Damages to the device caused by using other unmatched power adapters would not be covered in manufacturer's warranty service.

3 Product Overview

3.1 Appearance

The SVG401S SIP ATA is a next generation IP network device based on open industry standard SIP (Session Initiation Protocol). Built on innovative technology, the SVG401S SIP ATA features market leading superb voice quality and rich functionalities.

Power: Power adapter jack
FXS1 / FXS2 / FXS3 / FXS4: RJ11 port, FXS, Connect to regular analog phone or PABX
WAN: 1 RJ45 port. for Network connection
LAN: 1 RJ45 port. for PC/other networking equipment connection
Power: Power indicator.
REG: Server indicator. Registered: ON; Registering: Blinking; Do not register: OFF.

3.2 key Features

- Support two models: Bridge and Router(NAT&NAPT).
- Network Protocols: TCP/UDP/IP, ICMP, HTTP, DHCP Client (WAN Interface), DHCP Server(LAN Interface), DNS Client, DNS Relay, SNTP, PPPoE, FTP, TFTP
- VoIP Protocols: SIP (RFC3261, RFC3262, RFC3264, RFC3265)
- Voice Codecs: G.711(A-law/U-law), G.723.1, G.729A/B, iLBC
- NAT penetration: Support STUN client
- Support modify SIP register port, HTTP server port, Telnet server port and RTP port.
- Support two SIP server synchronously: Can register two different SIP server, and can make a call by either proxy.
- Support standard voice features such as numeric Caller ID Display, Do-Not-disturb, Forward, ban outgoing, call waiting, hold, transfer(blind transfer/attended transfer)
- Support digest authentication with MD5
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control).
- Provide easy configuration thru manual operation (Web interface and Telnet) or automated centralized configuration file via TFTP or HTTP.
- Support firmware upgrade via TFTP/FTP and HTTP.
- Support syslog, can send event of gateway to syslog server.
- Support reverse polarity

3.3 Hardware Specification

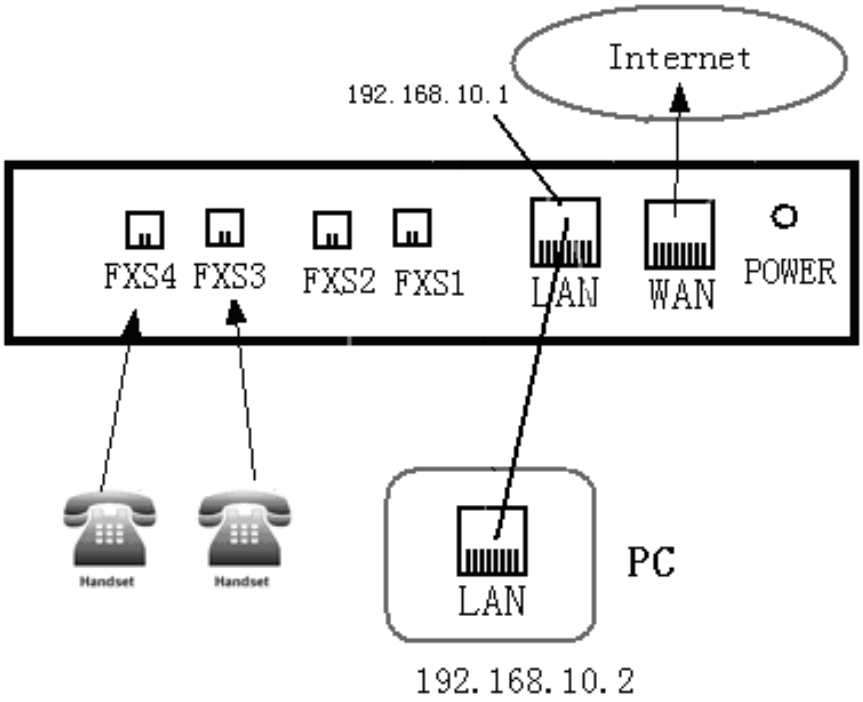
The table below describes the hardware specification.

Power Adapter	Input	110-220V AC
	Output	12V DC 2A
Port	WAN	1 10/100Base T RJ-45
	LAN	1 10/100Base T RJ-45

	FXS	4 RJ11 for phone
Power Consumption		2.8W/1.9W
Operating Temperature		0～60℃
Relative Humidity		5～95%
Volume		280mm*180mm*80mm
Weight		840g

4 Web configuration

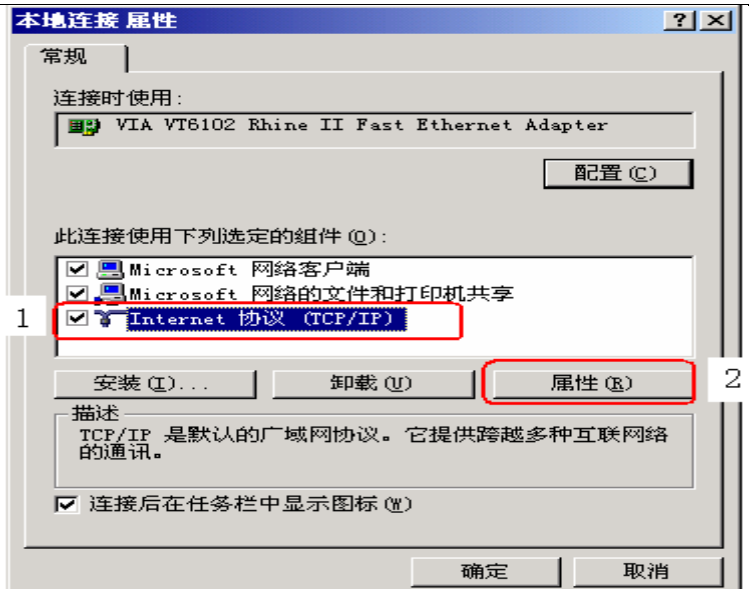
4.1 Physical connection



4.2 Preparation for Web configuration

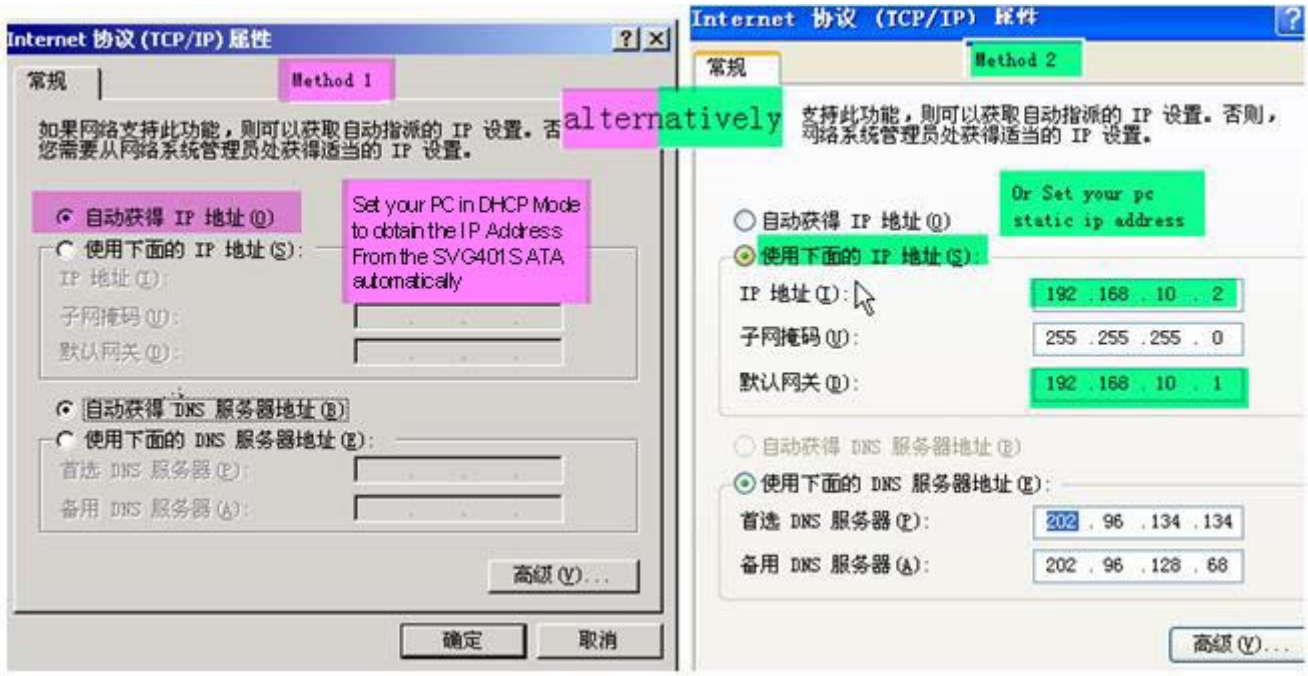
The SIP ATA Web Configuration Menu can be accessed by the following URL: http://Gateway-IP-Address. The default LAN IP address is “192.168.10.1” and WAN IP address is “192.168.1.179”. If the web login port of the gateway is configured as non-80 standard port, then user need to input http://xxx.xxx.xxx.xxx: xxxx/, otherwise the web will show that no server has been found. If connect PC with SVG401S LAN port and config to obtain IP address automatically, you could check the default gateway IP which is LAN IP address. The procedure is as below

a) Access to “Property of local area connection” dialog box



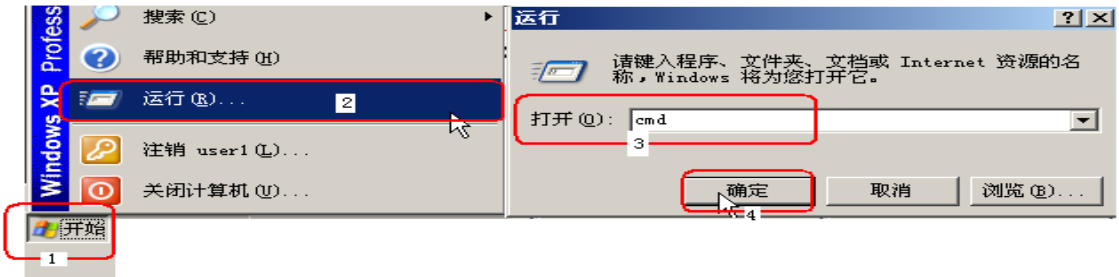
b) Select “Internet Protocol (TCP/IP)”,click “Property” button

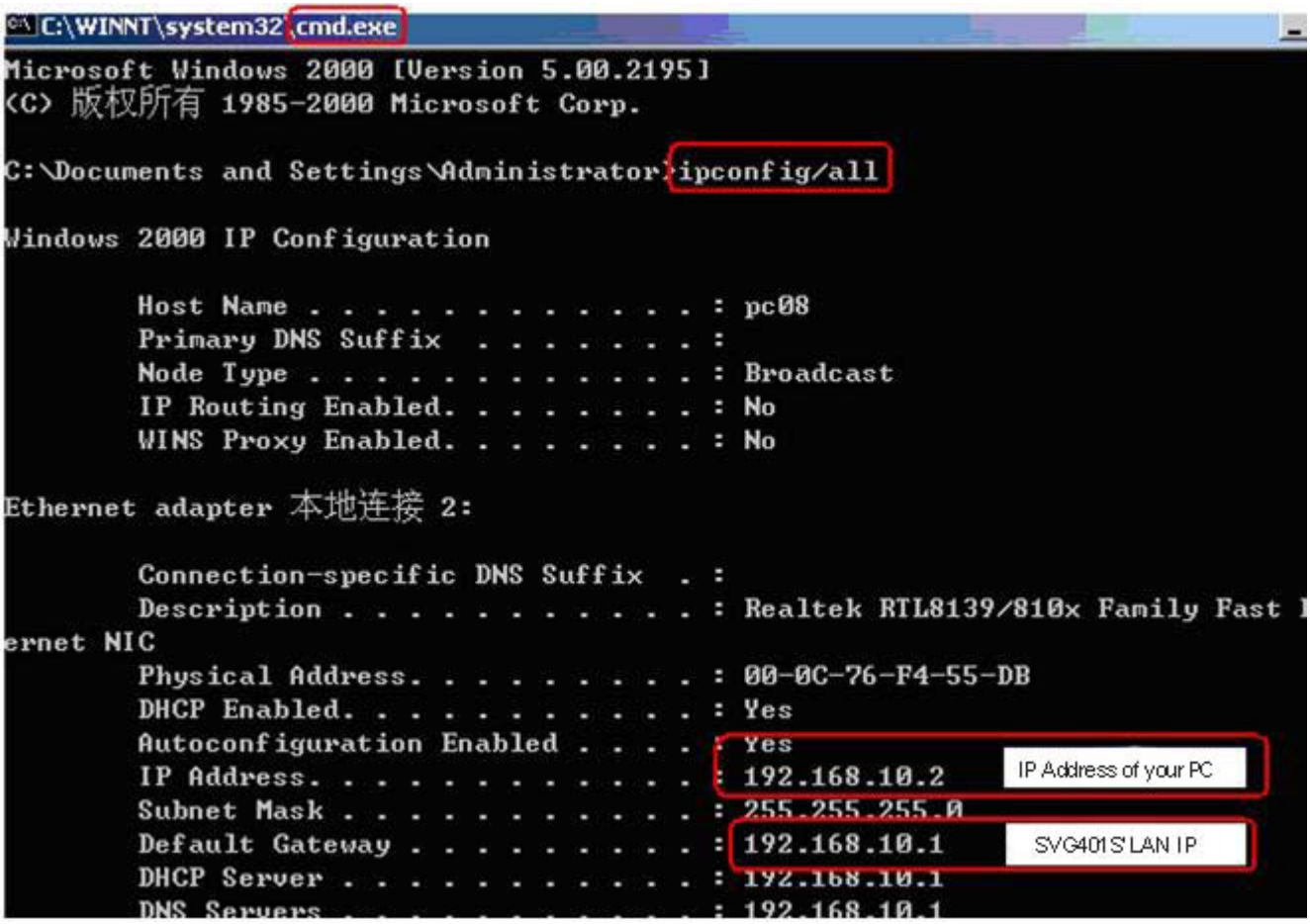
c) Setting refers to below dialog box, and then click “OK” button, PC will obtain IP address automatically.（or set your pc static ip 192.168.10.2）



Method 1 **Method2**

d) Input “cmd” command on the RUN submenu under PC START, key in “ipconfig/all on the command lines dialog box to find the default gateway IP address, which is the SVG401S LAN IP address

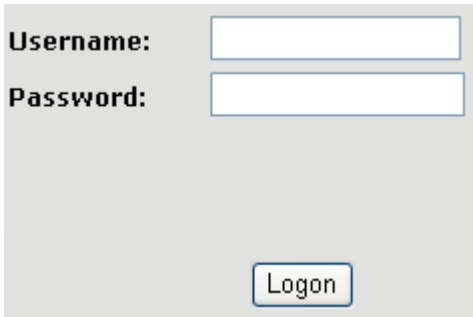



C:\WINNT\system32\cmd.exe
Microsoft Windows 2000 [Version 5.00.2195]
(C) 版权所有 1985-2000 Microsoft Corp.
C:\Documents and Settings\Administrator>ipconfig/all
Windows 2000 IP Configuration
Host Name . . . . . : pc08
Primary DNS Suffix . . . . . :
Node Type . . . . . : Broadcast
IP Routing Enabled. . . . . : No
WINS Proxy Enabled. . . . . : No
Ethernet adapter 本地连接 2:
Connection-specific DNS Suffix . :
Description . . . . . : Realtek RTL8139/810x Family Fast
ernet NIC
Physical Address. . . . . : 00-0C-76-F4-55-DB
DHCP Enabled. . . . . : Yes
Autoconfiguration Enabled . . . . : Yes
IP Address. . . . . : 192.168.10.2
IP Address of your PC
Subnet Mask . . . . . : 255.255.255.0
Default Gateway . . . . . : 192.168.10.1
SVG401S LAN IP
DHCP Server . . . . . : 192.168.10.1
DNS Servers . . . . . : 192.168.10.1

4.3 User verification

Users are requested to make verification when config or browse the SVG401S ATA through web page, users can direct login the config menu by inputting username and password as below ,

Default username and password is:

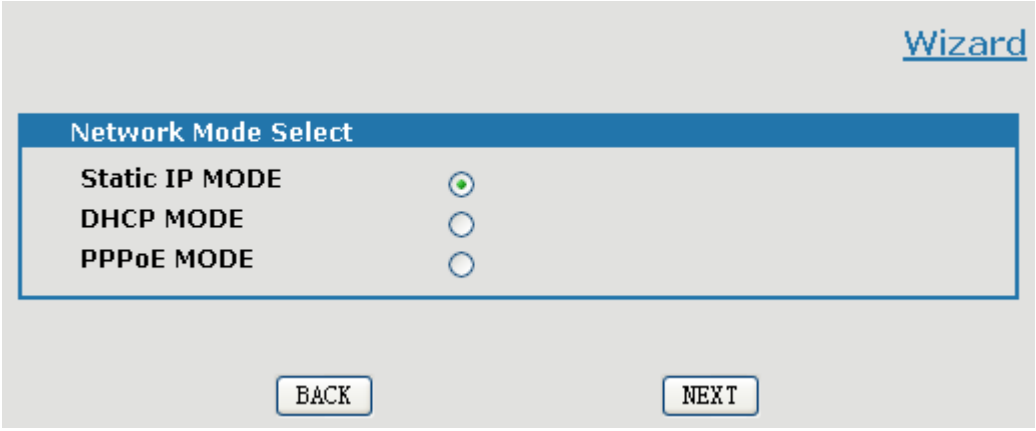


Username: admin password: admin highlevel user interface
Username: guest Username: guest low level user interface

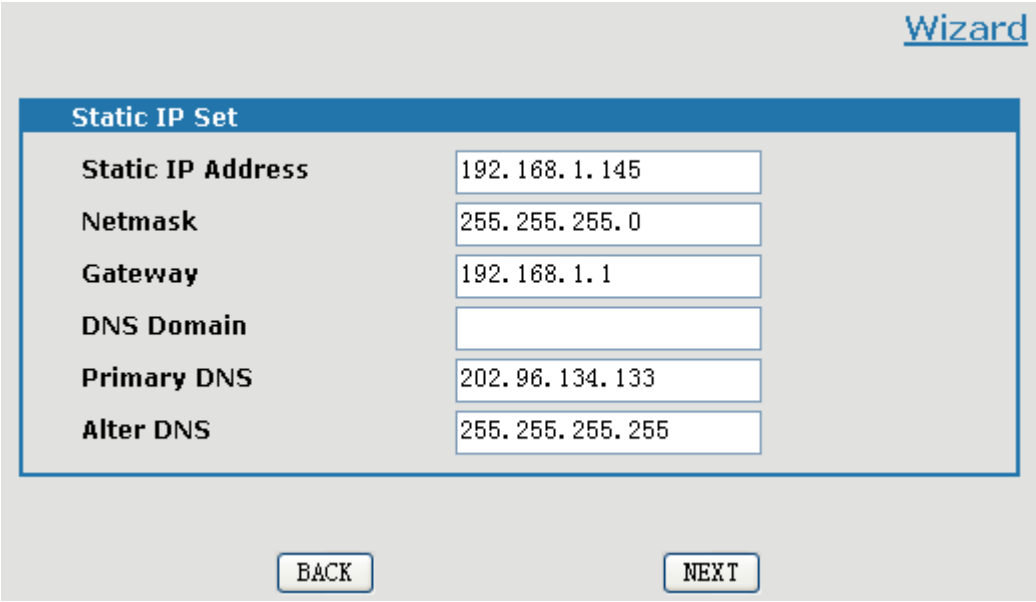
4.4 Wizard

In order to make user easier set SVG401S's parameters, the SVG401S provides a wizard flow to set step by step.

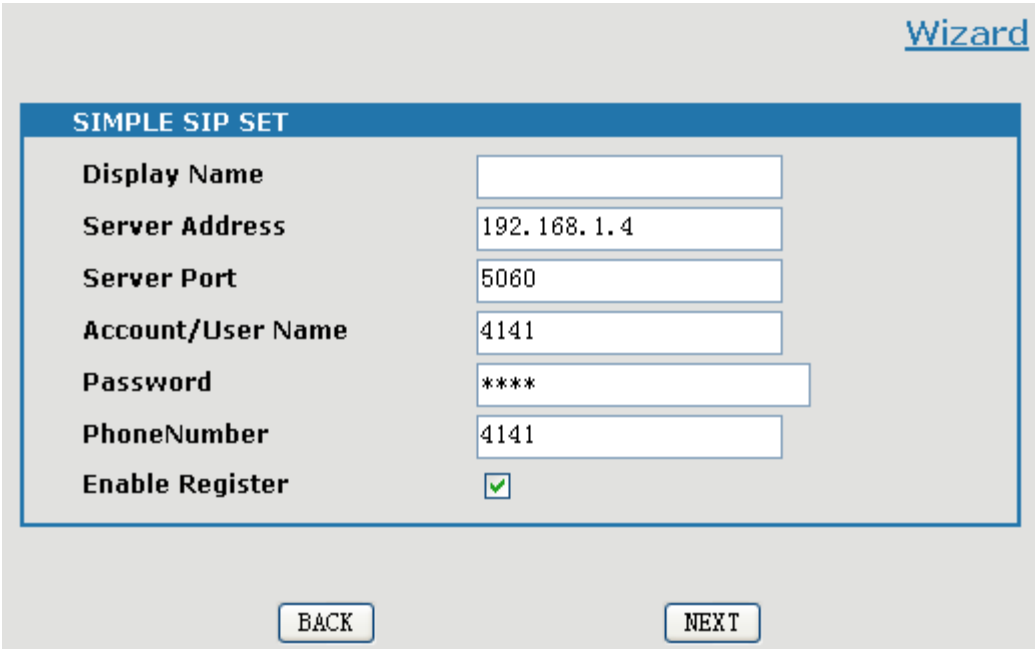
First step is selecting network mode



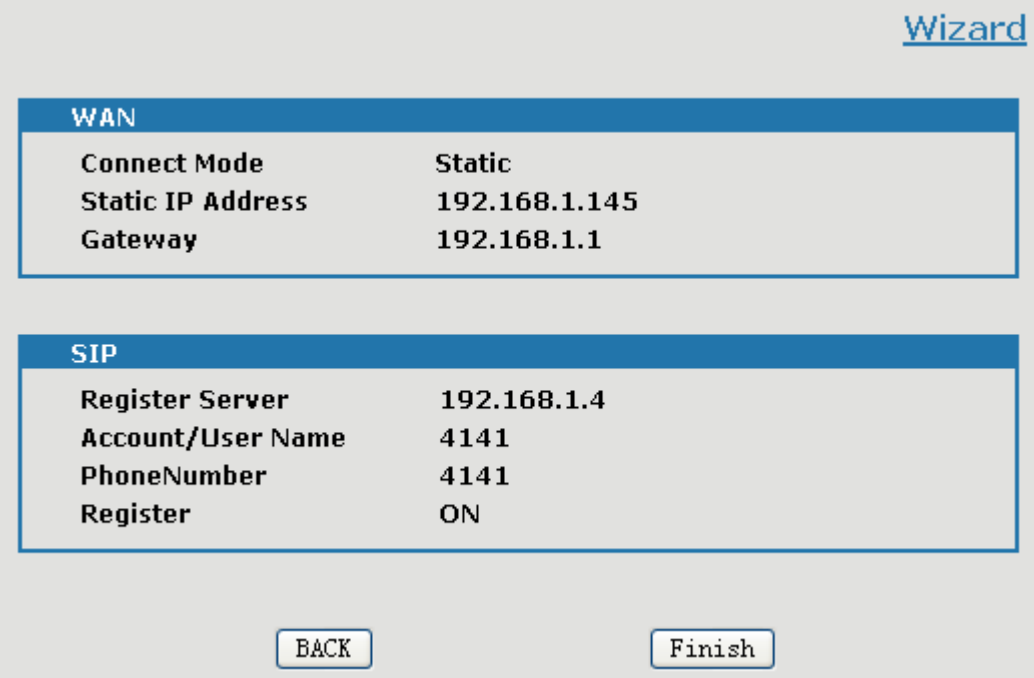
Second step is setting network parameters if user wants to set up Static IP or PPPOE mode, else jump to step 3 directly



Third step is setting SIP



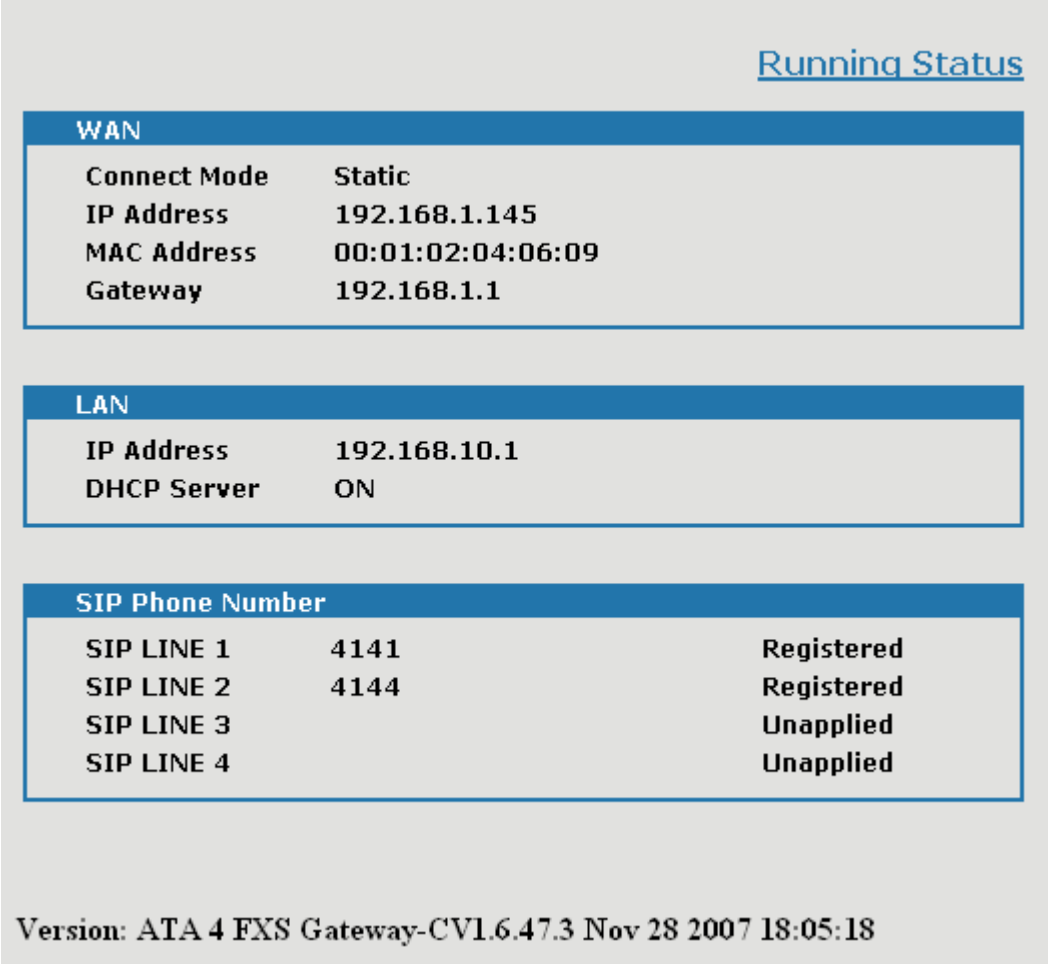
The last step is show the result of wizard. If user mistakenly configured the wizard, he could turn back to correct the configuration. Click Finish button, the SVG401S ATA will save the setting and reboot. Then, everything's set, users can start to make/receive VoIP calls.



4.5 Current State

On this page user can see the Running Status of the SVG401 VoIP ATA,

- Network section: Display the current WAN, LAN configurations of the SVG401S ATA
- Phone Number section: Display the phone number for each SIP line



- The version information is often shown at the bottom of this page

4.6 Network configuration

Network configuration includes WAN Config and LAN Config and other config correlative with network.

4.6.1 WAN Configuration

This web page displays the WAN parameter configuration.

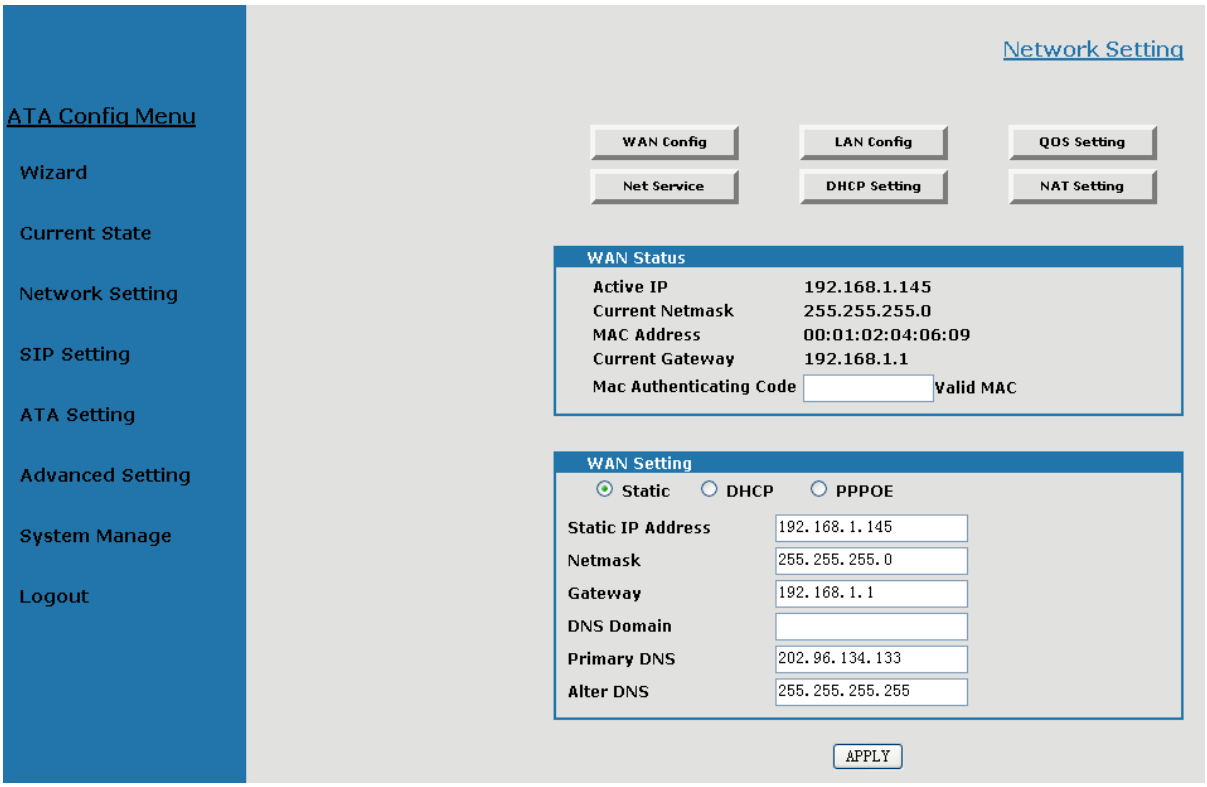
User can view the current network IP linking mode of the system on this page.

User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If DHCP is chosen, the SVG401S ATA will obtain IP address dynamically.

If PPPoE (adsl) is chosen, users need to set up a XDSL account to run the network, and obtain the IP address.

Note: In either ways, if IP address has been resigned, users need to access the web page with the new IP Address.

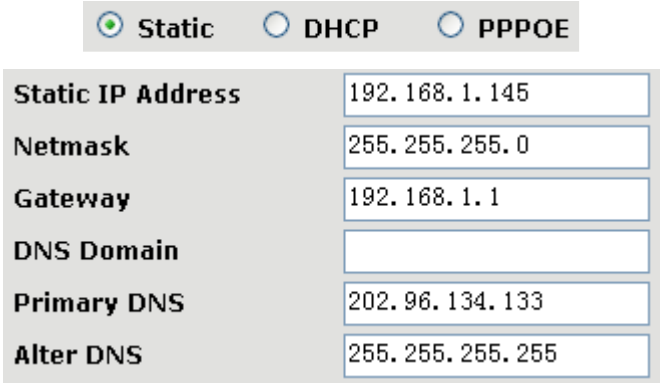


Display <valid MAC > means the SVG401S had been certificated.

Display <invalid MAC> means the SVG401S ATA needs a MAC Authentication Code. (You can get it from CHIMA), in this way, the SVG401S would not work properly.

- **WAN Port static mode configuration**

By default, the network mode is DHCP; When setting Static IP mode, Users need to set below parameters



IP Address	WAN IP address
Netmask	Network mask
Gateway	Default gateway IP address
DNS Domain	Optional configuration
Primary DNS	IP address for primary Domain Name Server
Alter DNS	Optional configuration

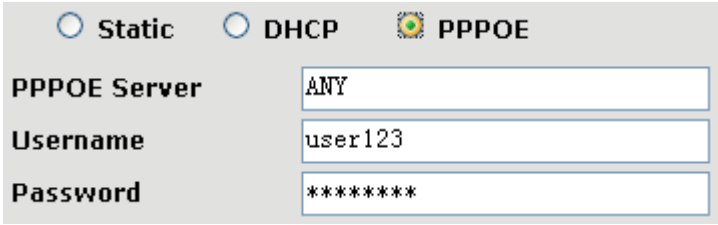
Click “Apply” button after finished above settings, the SVG401S ATA will save the settings automatically, and effect immediately. If users need to access the SVG401S WEB Configuration Page through WAN, the WAN IP Address should be found out first. (IVR: Dial #*111# to hear the SVG401S IP Address)

- **WAN port DHCP mode configuration**

Select “DHCP”, the SVG401S ATA will auto-config the WAN parameters, and effect immediately.

- **WAN port PPPoE mode configuration**

Select “PPPOE”, users need to set up a Xdsl account which was provided by the ISP

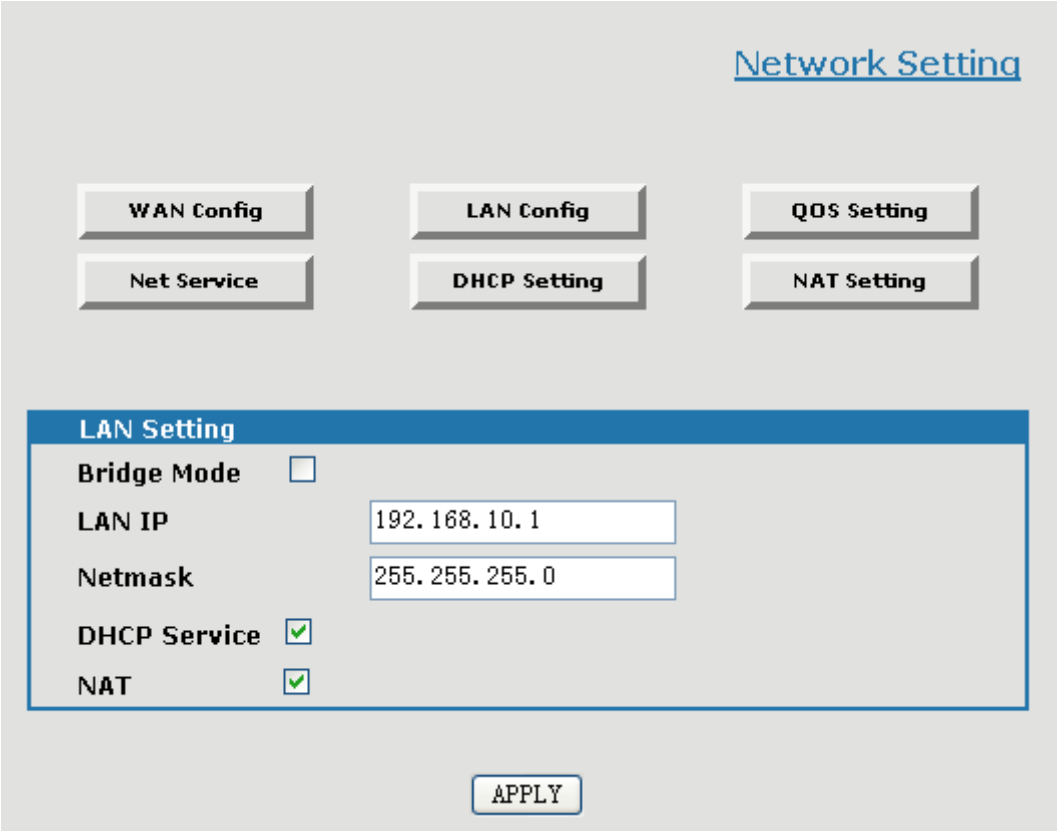


Server	Remain the default value If ISP doesn’t require special configuration
User	Provided by User’s ISP
Password	Provided by User’s ISP

Click “Apply” button after finished above settings, the SVG401S ATA will auto-config the WAN parameters and effect immediately.

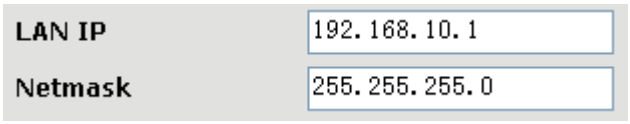
4.6.2 LAN Configuration

Click LAN Config button to show LAN settings. This web page displays the LAN parameters. Please note that, once the bridging mode is selected, the LAN configuration will be no longer effective.



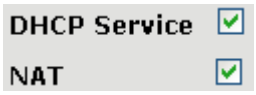
Configuration Example

- Config LAN: config one private IP address



IP	LAN IP address
Netmask	Network Mask

- **DHCP and NAT selections, by default, the NAT is enabled.**

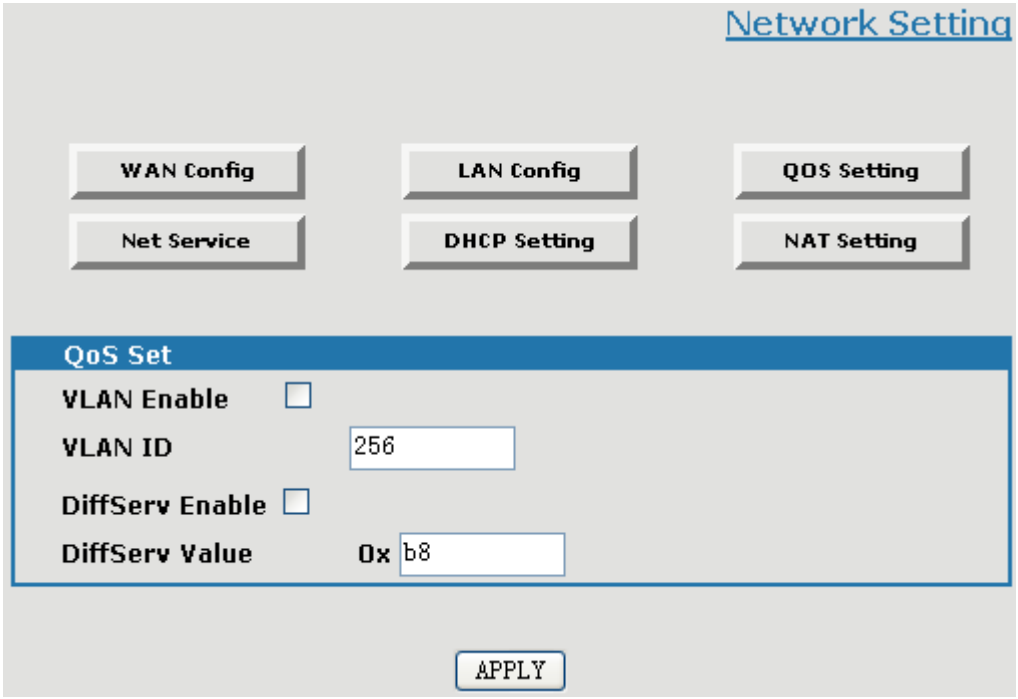


Start Bridge Mode or not (transparent mode): Once Bridge Mode is enabled, some parts of LAN configuration will be disabled, and the ATA will no longer set IP address for LAN physical port, LAN and WAN then will be in the same network;

4.6.3 QOS Configuration

SVG401S supports DiffServ and 802.1Q/P QOS modes to ensure the voice quality.

Click QOS button to show the QoS settings.



User needs to set SVG401S in bridge mode before configuring the VLAN.

VLAN can be enabled or diabled.

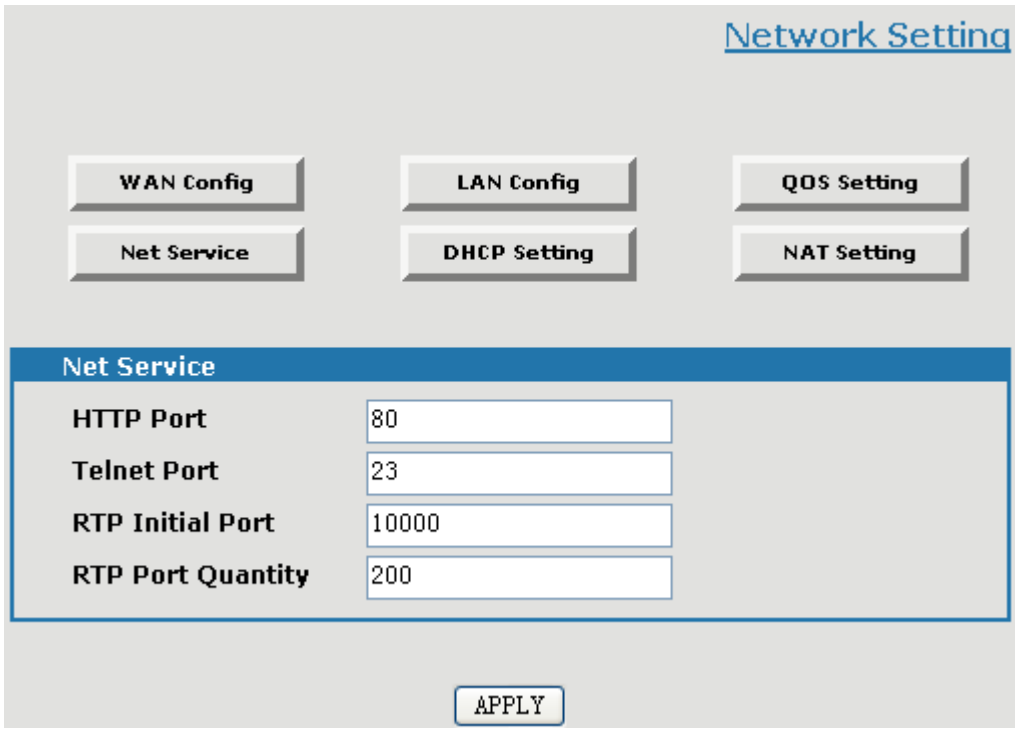
VLAN ID is the parameter that VLAN manager sends to user to ensure the ATA is in correct VLAN.

DiffServ Enable is to enable or disable QOS

DiffServ Value is Hex value to set different priorities for voice packets.

4.6.4 Net Service

Click Net Service buttone to show service ports of SVG401S ATA.



HTTP Port is the configuration port of web configuration access, by default, it is port 80, if you want to enhance system safety, you can change it into non-80 standard port.

Example: The ip address is 192.168.1.70, you change the port value to 8090, the accessing address of the ATA will then be in such format: http://192.168.1.70:8090

But if the value is 0 , the ATA would not be accessible by web browser.

Configure telnet port, by default, the port is 23. You can change the port number.

Example: The ip address is 192.168.1.70, you change the port value to 8023, the Telnet accessing address of the ATA will then be in such format: telnet 192.168.1.70:8023

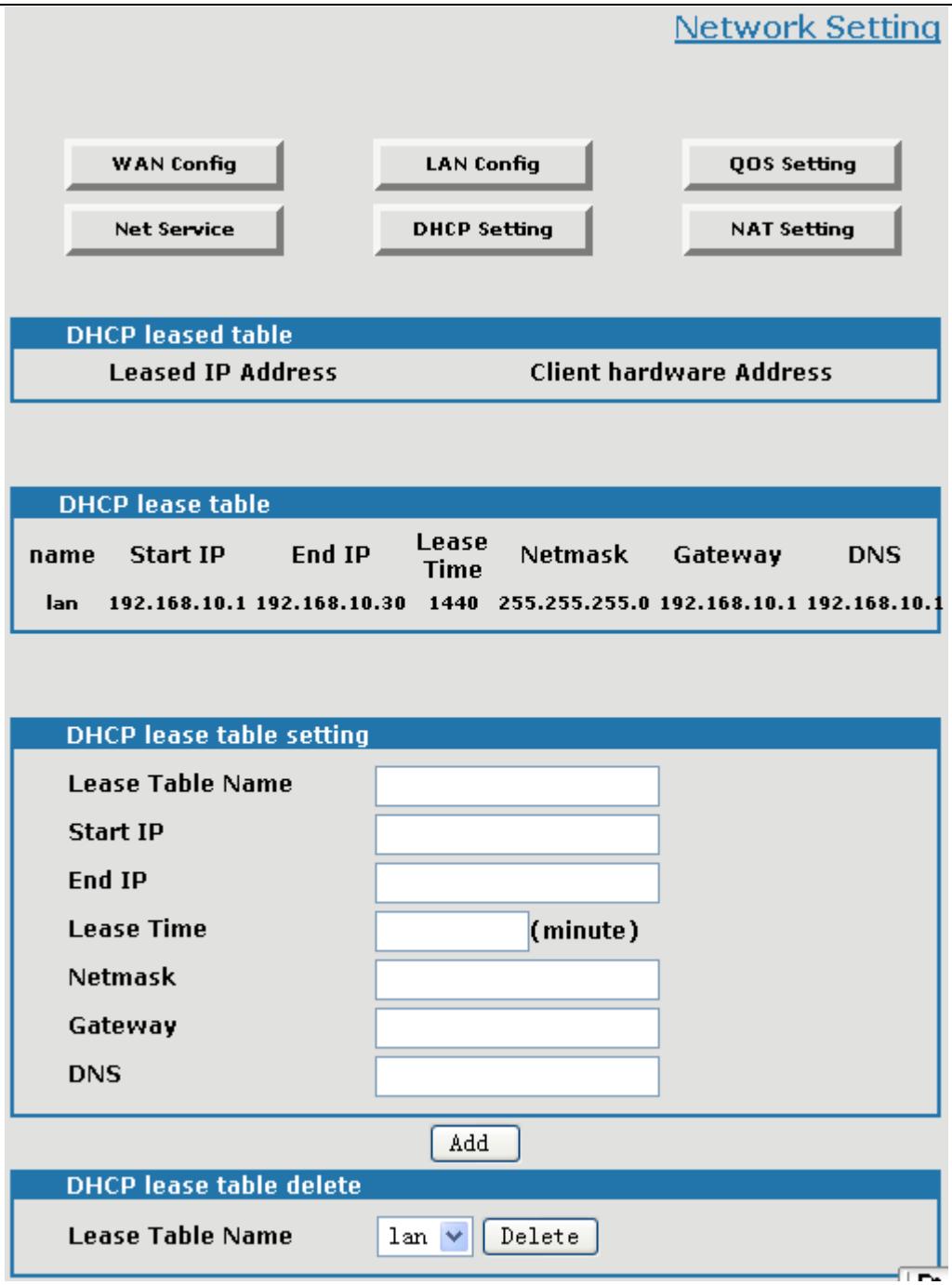
RTP initial port is SVG401S's voice port which is used to send RTP packets. It is dynamic allocation.
Configure the maximum quantity of RTP port. The default value is 200;

The configuration on this page should be saved after changing, and it will be effected after reboot. To change the HTTP or Telnet Port number, it's suggested to set a port number greater than 1024 for the system will save port number which is greater than 1024.

4.6.5 DHCP Setting

Click DHCP Setting button to show SVG401S's DHCP parameters.

When SVG401S work as a router, This setting is for network device connects to SVG401S's LAN port.



DHCP leased table is the Leased IP-MAC correspondence table of DHCP. The table will display all the devices which are assigned an IP address by SVG401S router.

DHCP lease table shows the current table of LAN port. It can be set more than 1 table. SVG401S would assign DHCP IP addresses from different tables.

User may use below setting to add a new lease table.

Lease Table Name: Lease table name.

Lease Time: DHCP server lease time.

Start IP: Start IP of lease table.

End IP: End IP of lease table.

Netmask: Netmask of lease table.

Gateway: Default gateway of lease table

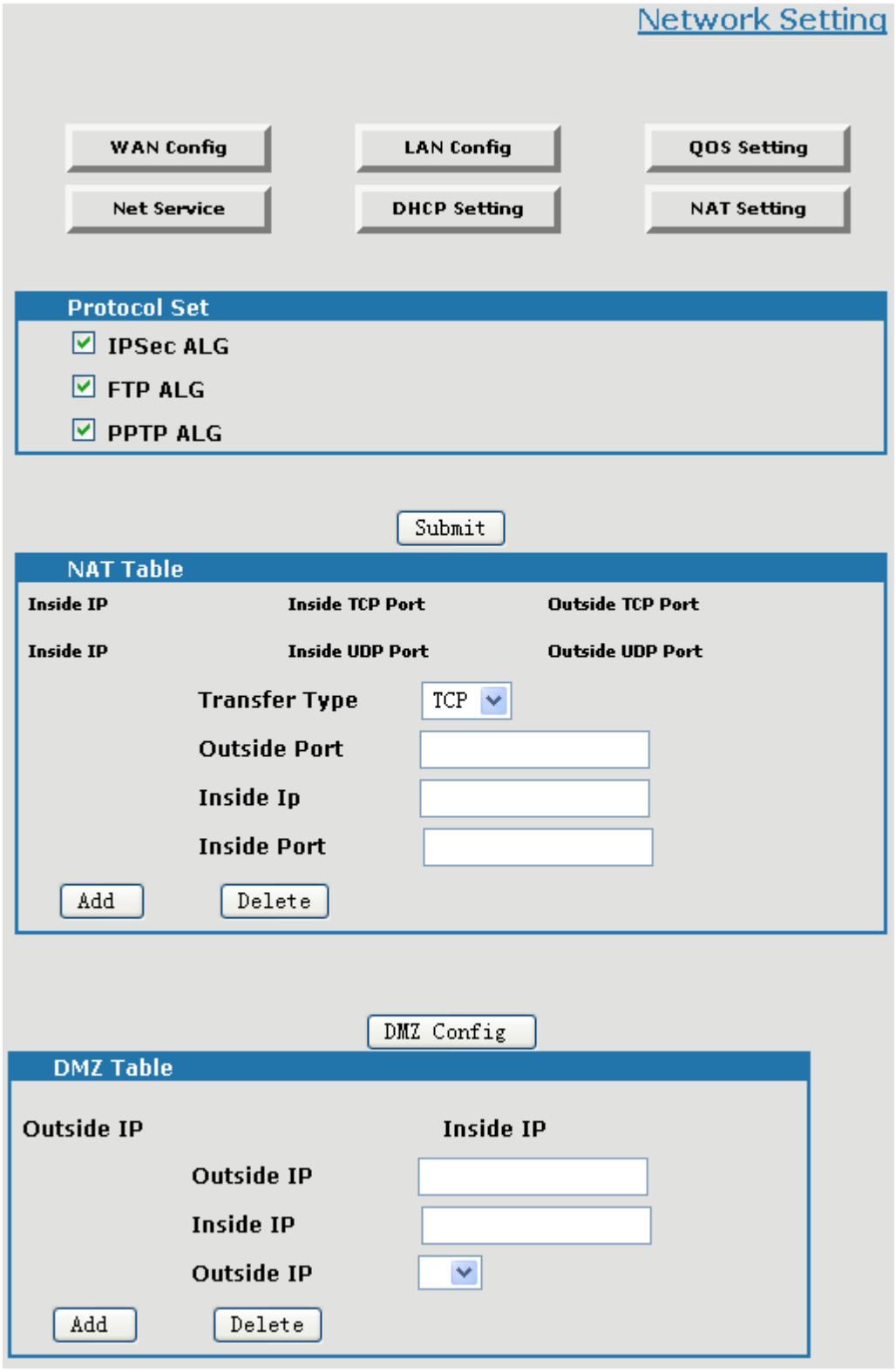
DNS: Default DNS server of lease table.

Notice: This setting won't effect unless you save the configuration and reboot the device

4.6.5 NAT Setting

This page is for NAT configuration ,such port forward ,DMZ .

Network Address Translation (NAT) provides a mechanism for a privately addressed network to access registered networks, such as the Internet, without requiring a registered subnet address. This eliminates the need for host renumbering and allows the same IP address range to be used in multiple intranets. With NAT, the inside network continues to use its existing private or obsolete addresses. These addresses are converted into legal addresses before packets are forwarded onto the outside network.



Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

IPSec ALG: Enable/Disable IPSec ALG;
FTP ALG: Enable/Disable FTP ALG;
PPTP ALG: Enable/Disable PPTP ALG;
Transfer Type: Transfer type using port mapping.
Inside IP: LAN device IP for port mapping.
Inside Port: LAN device port for port mapping.
Outside Port: WAN port for port mapping.

Click Add to add new port mapping item and Delete to delete current port mapping item.

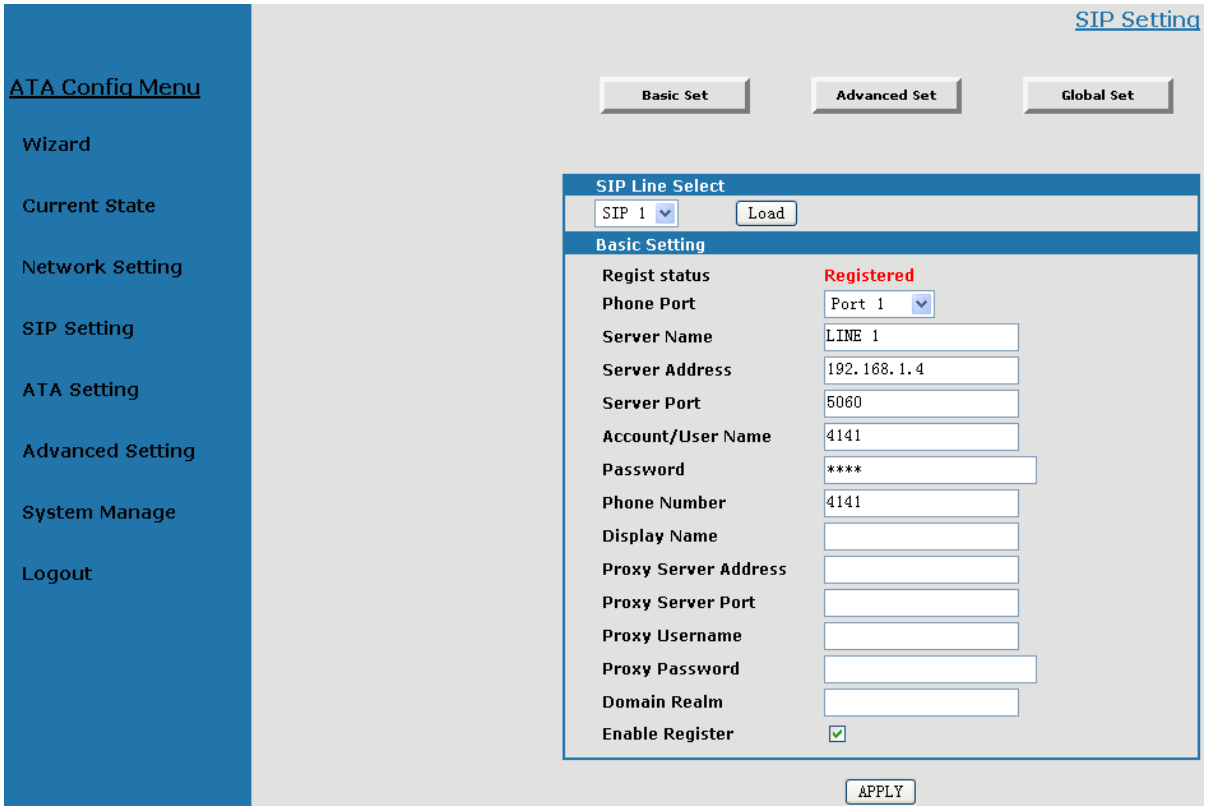
Click DMZ Config to show DMZ configuration. Setting outside IP and Inside IP, then click Add button, other device of WAN would have access to the device which you enabled DMZ for.

4.7 SIP Setting

This section is to config signaling protocol for the SIP Server and Client.

4.7.1 SIP Line basic configuration

User can configure specific parameters of SIP line on this page;
Click Basic to set the common parameters of SIP line selected.

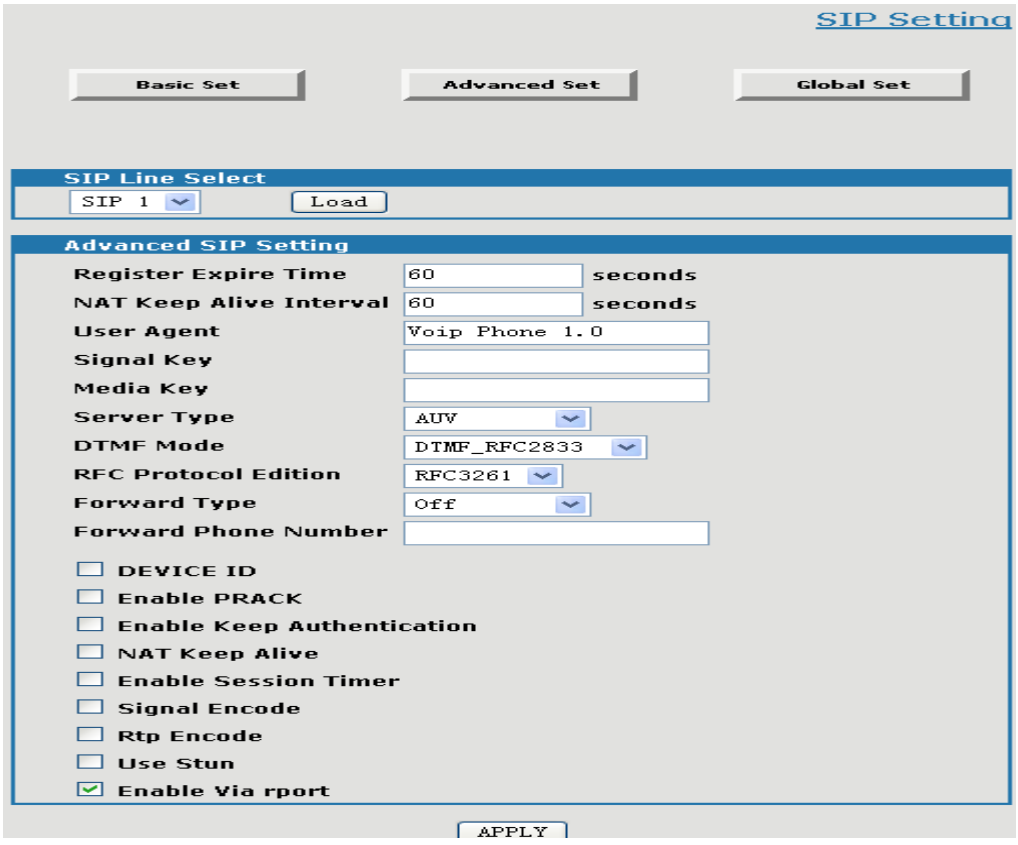


Definition of each parameter described as below

SIP Line Select	Select a line of SIP to configured, then click load button to get ready to set paramters
Regiseter status	SIP register state, if register successfully, show “Registered，otherwise show different results
Phone port	Choose the corresponding port of SIP line
Server name	Name the SIP Line with a customized name
Server address	Set SIP server IP address or Domain
Server Port	Set SIP server port
Account/User Username	Set SIP account username
Password	Set password of SIP server account
Phone Number	Set assigned phone number
Display Name	Set assigned SIP display name
Proxy Server Address	Set proxy server if SIP server requires proxy settings
Proxy Server Port	Set proxy server port
Proxy Username	Set proxy server account username
Proxy Password	Set password of proxy server account
Domain Realm	Enter the sip domain if any, otherwise the SVG401S will use the proxy server address as sip domain. (Usually it is same with registered server and proxy server IP address).
Enable Register	After setting all the parameters above, enable register and click apply to make SVG401S register to SIP server

Click Advanced set button to set the other parameters.

4.7.2 SIP Line advanced configuration

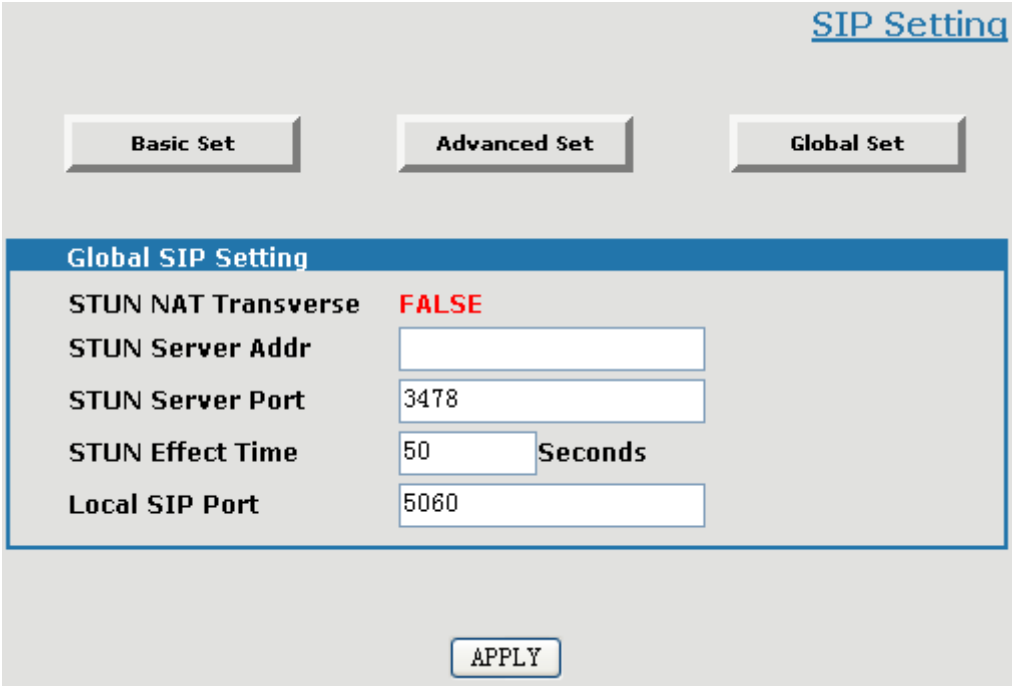


Definition of each parameter described as below

Register Expire Time	Set expire time of SIP server registration, the final actual time maybe decided by server
NAT Keep Alive Interval	Set NAT keep alive interval time to make the SVG401S in LAN keep being contacted if server don't support keeping NAT alive
User Agent	Set the user agent name to show the information of the SVG401S ATA
Signal Key	Fill in the key for SIP encryption according to the Server type selected. Default is common
Media Key	Fill in the key for Media encryption according to the Server type selected. Default is common
Server Type	Set SIP server type
DTMF Mode	Set DTMF sending mode, support RFC2833, DTMF_RELAY (inband audio) and SIP info
RFC Protocol Edition	Enable the phone to use protocol edition. By default, the ATA uses RFC3261, by RFC protocol edition, it can change to RFC2543, etc.
Forward Type	Set the selected Line's call forward function
Forward Phone Number	Set the selected Line's call forward number
DEVICE ID	Enable/Disable to use a special way to register the SVG401S to a special server, which needs to use device id.
Enable PRACK	Enable / Disable SIP prack function. Default is unchecked. Prack is same with ACK that is used for temporary response. Prack is one of the SIP request info
Enable Keep Authentication	Enable /disable registration with authentication, so that the SVG401S doesn't need to re-fill the authentication field each time. When server receive the register request with authentication, server will response 200 message, not 401
NAT Keep Alive	Enable/disable NAT keep alive function
Enable Session Timer	Enable/disable support RFC4028.
Signal Encode	Enable/disable singal encryption according to the server type.
Rtp Encode	Enable/disable Voice encryption according to the server type
Use Stun	Enable/Disable STUN if SIP server does not support NAT
Enable Via rport	Enable/disale RFC3581, When the SVG401S is behind a NAT. "rport " allows it to request the server to send the response back to the IP address and port from which the request originated.

Click Global Set button to set the parameters efficiently for all SIP lines.

4.7.3 SIP Line global configuration



Definition of each parameter described as below

STUN NAT Transverse	Indication that current NAT is type that could be transverse or not
STUN Server Addr	Set the STUN server's IP address/domain
STUN Server Port	Set the STUN server's port
STUN Effect Time	Set the interval time of STUN detection
Local SIP Port	Set SIP signal port

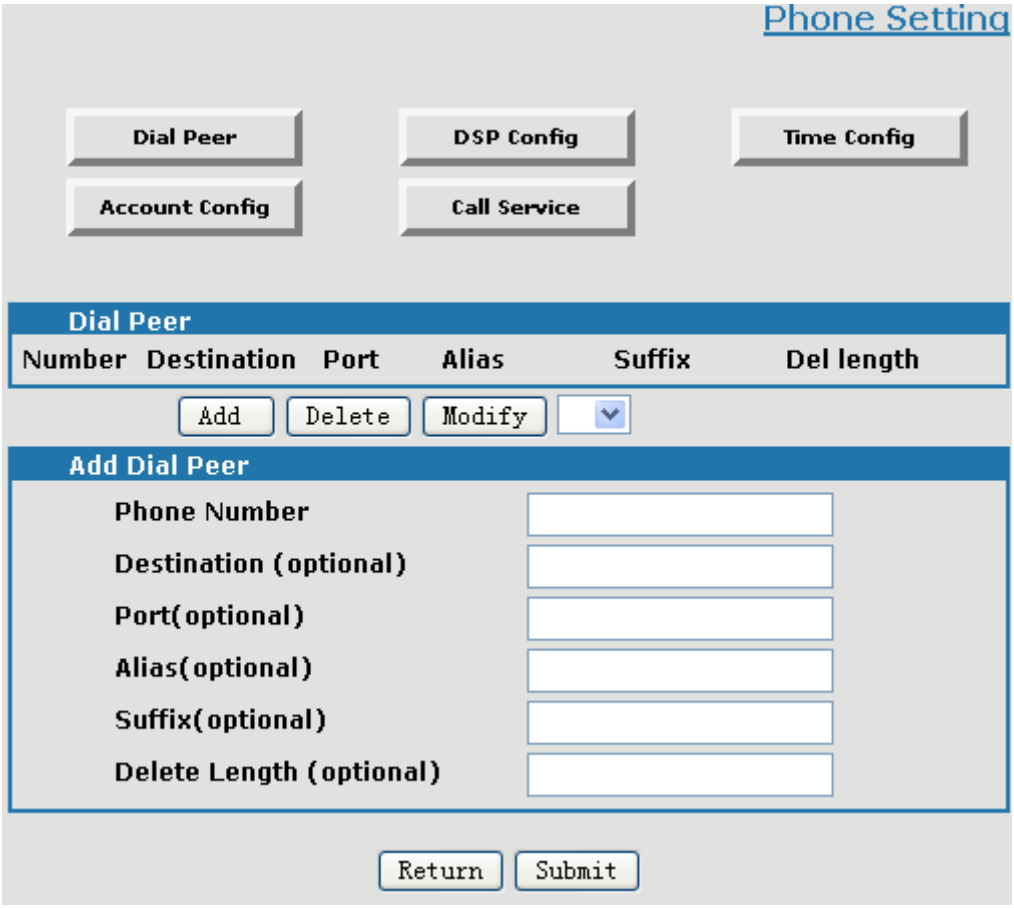
4.8 ATA Setting

4.8.1 dial peer configuration

Number IP table configuration

Function of number IP table is one way to implement the call online, and the call will be more flexible by configuring the number IP table. For example, user knows the other party's number and IP and want to make direct call to the party by point-to-point mode: the other party's number is 1234, make a configuration of 1234 directly, then the ATA will send the called number-1234 to the corresponding IP address; Or set numbers with prefix matching pattern. For example, user wants to make a call to a number in a certain region (010), user can configure the corresponding number IP as 010T—protocol-IP, after that, whenever user dial numbers with 010 prefix(such as 010—62201234), the call will be made by this rule.

Base on this configuration， we can also make the ATA use different accounts without manual swap. When making deletion or modification, select the number first and click Modify button to change the rule.



Click Add to add the Dial Peer configuration. It will be shown at the upper part of the page.

Phone Number is to add outgoing call number, there are two kinds of outgoing call number setup: One is exactitude matching: after this configuration has been done, when the number is the same with the user's dialing number, the ATA will make the call with this number's IP address image or configuration; Another is prefix matching（be equivalent to PSTN's district number prefix function, if the previous N digit of this number are the same with that of the user's dialing number(the prefix number length)， then the ATA will use this number's IP address image or configuration to make the call. When configuring the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching.

Destination(optional) is configuration of destination address，if it is point-to-point call，then input the opposite terminal's IP address, it can also be set as domain name and resolved the specific IP address by DNS server of the phone. If no configuration has been made, then the IP will be considered as 0.0.0.0. This is an optional configuration item.

Port(optional) is the callee's protocol signal port, this is an optional item; if nothing is input, the SVG401S will set it as the default value 5060 of sip protocol.

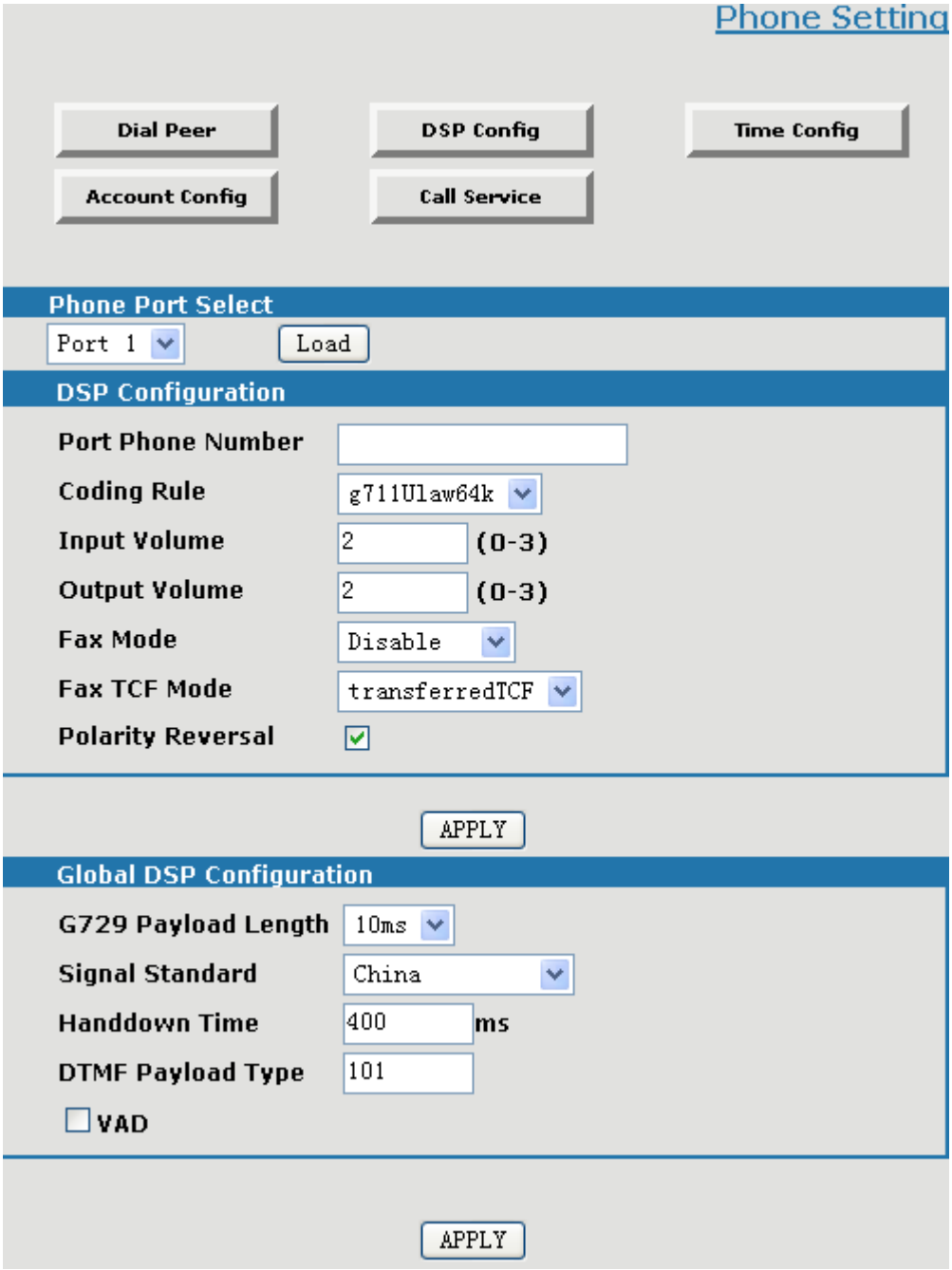
Configure alias，this is an optional item: it is the number to be used when the other party's number has prefix; when no configuration has been made, SVG401S will be shown as no alias.

add:	xxx	add xxx before number. in this way it can help user save the dialing length;
all:	xxx	the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all" the actual calling number will be replaced;
del		delete n bit in the front part of the number, n can be decided by the replacing length this configuration can decide the protocol for appointed number
rep:	xxx	n bit in the front part of the number will be replaced. n is decided by the replacing length.

Configure suffix, this is an optional item; it is the additive dial-out number behind the number; when no configuration has been made, SVG401S will be shown as no suffix.

4.8.2 DSP configuration

On this page, user can set speech coding，IO volume control, cue tone standard, caller ID standard and so on.



First, choose the phone port that you want to configure.

Port Phone number is the extension number which is applied by device ID.

- Configure Coding Rule according to network bandwidth and VoIP server; support G.711(A-law/U-law), G.723.1, G.729A/B, iLBC (G723high/low is optional which is support in special firmware version).

Input Volume is to adjust mike volume. The value is from 1 to 5.
Output Volume is to adjust handset out volume. The value is from 1 to 9.

FAX mode list include Disable fax function ,bypass-a bypass-b ,relay-t38 mode.
There are Two T.38 fax TCF modes for selection:
Local TCF means TCF do not transfer between gateways; TCF is generated locally
Transferred TCF means TCF and other picture data transfer data between two ATA. The current ATA supports this method.

Enable/disable reverse polarity for billing meters.

Normally, G729 Payload Length does not need to be changed into 10 ms; the voice payload size can be represented in terms of the codec samples. For example, a G.729 voice payload size of 20 ms (two 10 ms codec samples) represents a voice payload of 20 bytes [(20 bytes * 8) / (20 ms) = 8 Kbps]

Configure Signal Standard according to country's phone voice. It acts on the tone when you pick up phone, ring back tone and so on.

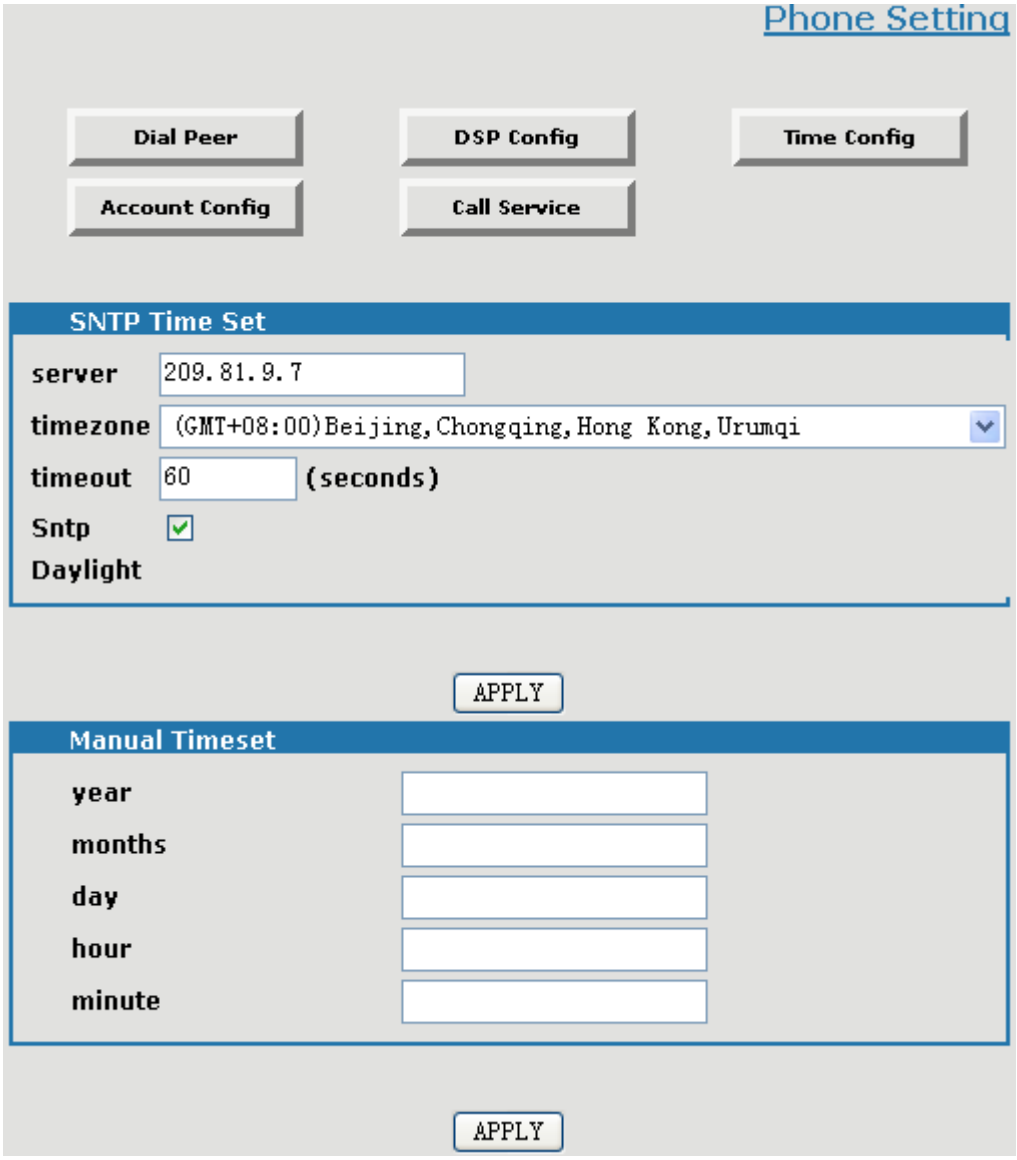
Adjust the handdown time .The time range is from 80ms to 800ms.

DTMF payload Type is when using 2833 as DTMF, its payload type is defined with different value by different ITSP's devices. So it should be set if ITSP requires to.

VAD Enable/disable VAD(Voice activity detection).

4.8.3 Time configuration

Setting time zone and SNTP (Simple Network Time Protocol) server according to users location, users could manually adjust date and time on this web page.



Definition of each parameter described as below

Server	**SNTP server IP address**
Time zone	Set time zone according to users location, Beijing time zone is GMT:+8:00
Timeout	use for socket calls which will block while doing the SNTP query, default value is 60 seconds
Manual Time set	Manually set date and time

4.8.4 Account configuration

Users can edit users (add or delete) account and modify existing users' authority on this web page.

Definition of each parameter described as below

User Name	List existing phone user account name
User Level	Show existing user account level [two option]:Root and General: Root level users have the right to modify configuration; General level users have the right to read-only
Add	Add user account to IP phone
Delete	Delete increased user account
Modify	Modify increased user level and password

Operation Example:

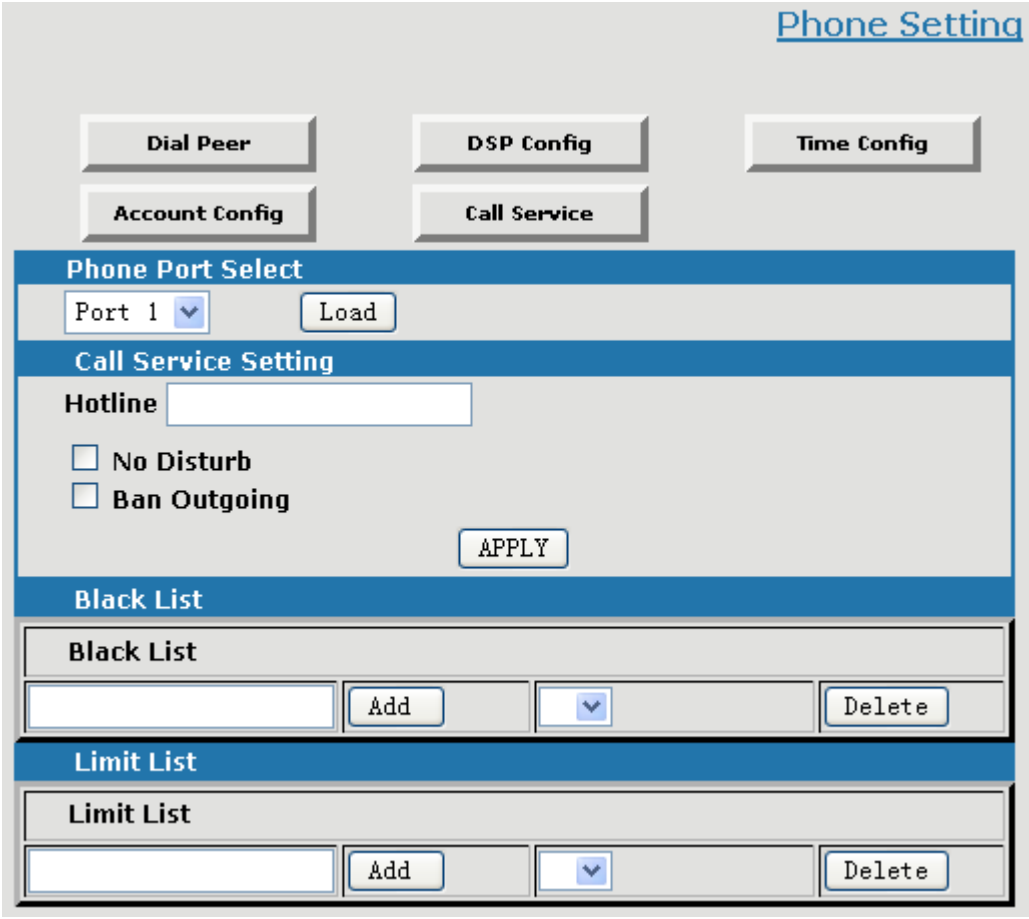
Add one new account:
Click “Add” button →input User name (No-Modify) →Choosing User level from dropdown menu →set new user password →confirm password →submit the new account info by clicking “submit” button →show “submit success” on screen →return to account configuration interface by clicking “Return” button

Delete increased account:
Choosing the account need to del from dropdown menu→ Delete account by pressing “Delete button” →show “Submit Success” on screen

Modify account’s password:
Choosing the modified account →enter below interface →modify user password →click “Submit” button to submit the modification

4.8.5 Call Service configuration

On this page, user can set value added services such as hot-line, blacklist, out-limit list and so on.



First, choose the phone port need to set.

Configure hot-line number of the port. With this number of the port， this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number. If you do not use hotline , please let it be blank.

If No Disturb is enabled, the phone will not ring when there is a incoming call. DND, do not disturb, enable this option to refuse any calls.

When ban outgoing is enabled for the port, the port’s phone will not make calls, but it can receive calls.

Incoming call in these phone numbers will be refused. It is for precluding incoming communication like Call ID. If user don't want to answer a certain phone number, please add this phone number to the list, and then this number will be unable to get through the phone.

Outgoing calls with these phone numbers will be refused. for example, if user don't want the phone to dial a certain number, please add the number to this table, and the user will be unable to get through this number.

SVG401S also support call hold/waiting/transfer.

Users can press flash keypad or flash hooks of phone, so the party you are talking will be hold; press flash keypad or flash hooks of phone again, the call will be resumed.

When talking with some person, another one calls in, users

will hear waiting tone to remind user there is a new call in. User could press the flash keypad or flash hooks to hold the current call and accept the new coming call; user can also press the flash keypad or flash hooks to switch the calls between two parties.

SVG401S can use blind transfer and attended transfer function.

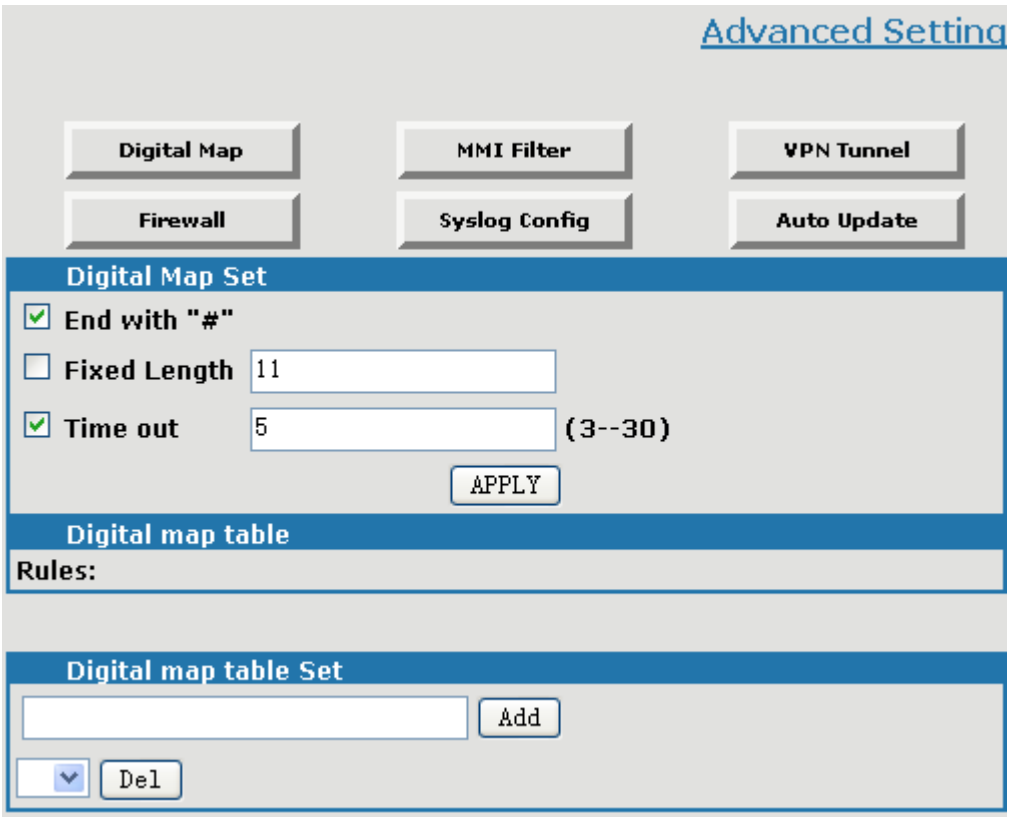
When the caller calls SVG401S, SVG401S accepts. Then SVG401S user presses the flash keypad or flash hooks, pressing *number#, then hangs up. The call will then be transferred(blind transfer).

When the caller calls SVG401S, SVG401S accepts. Then SVG401S user presses the flash keypad or flash hooks, then dial C. After talking, SVG401S hangs up. The call will be transferred(attended transfer) to C.

4.9 Advanced Setting

4.9.1 Digital Map configuration

Digit map is a set of rules to determine when the user has finished dialing. Digital Map is based on some rules to judge when user end their dialing and send the number to the server. With digital map, users don't have to press'#' key or "call" key after finish dialing. If the number dialed matches some item in the digital map table, or it doesn't match with any item, this number will be sent out immediately. It is not like using dial peer; using digital map won't change the number dialed, the number sent is the same as the number dialed.



End With "#": Checked- use # for the end of dialing. User dials 200300#. phone send 200300 to server immediately.
Unchecked-dial # as a digital number. Example: User dial #1966, sever receive #1966.

Fixed Length: when the length of the dialing is matched, the call will be sent.

Timeout: Specify the timeout of the last dial digit. The call will be sent after timeout.

End with "#", fixed length and Time out, these options are not repulsion. That means, these three ways can be used together or alone.

User could also use digital map table to make rules.

X Represents any numbers between 0 and 9.

Tn Represents the last digit timeout. Here [n] represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

. (Dot) represents any number and no length limit.

[ ] Number location value range. It can be a number range(such as [1-4]), or number is separated by comma such as [1,3,5], or use a list such as [234]

Example:

[1-8]xxx any 4 digits number between 1000 and 8999 sending out immediately.

9xxxxxxx any 8 digits number starting with 9 will be sent out immediately.

911 after finishing dialing 911, it will be sent out immediately.

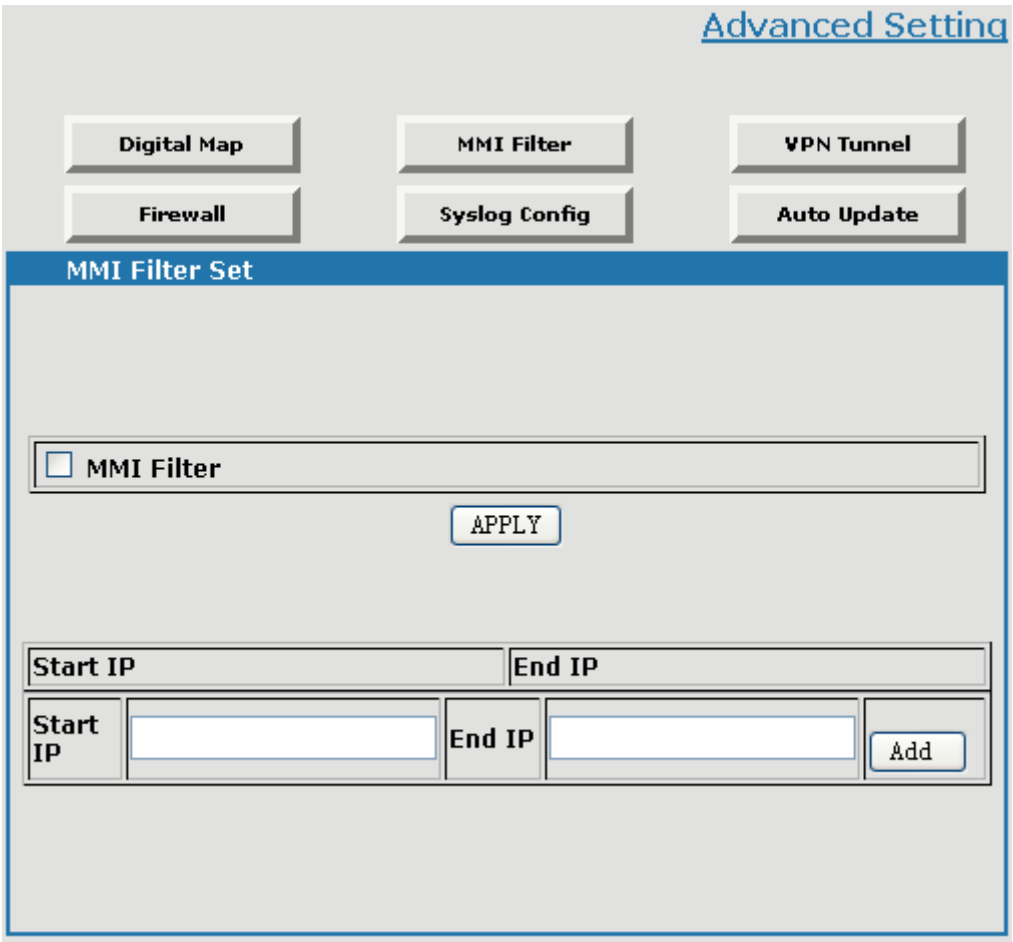
99T4 after finishing dial 911, it will be sent out in 4 second.

9911x.T4 any number more than 5 digits length starting with 9911, will be sent out in 4 seconds.

4.9.2 MMI Filter configuration

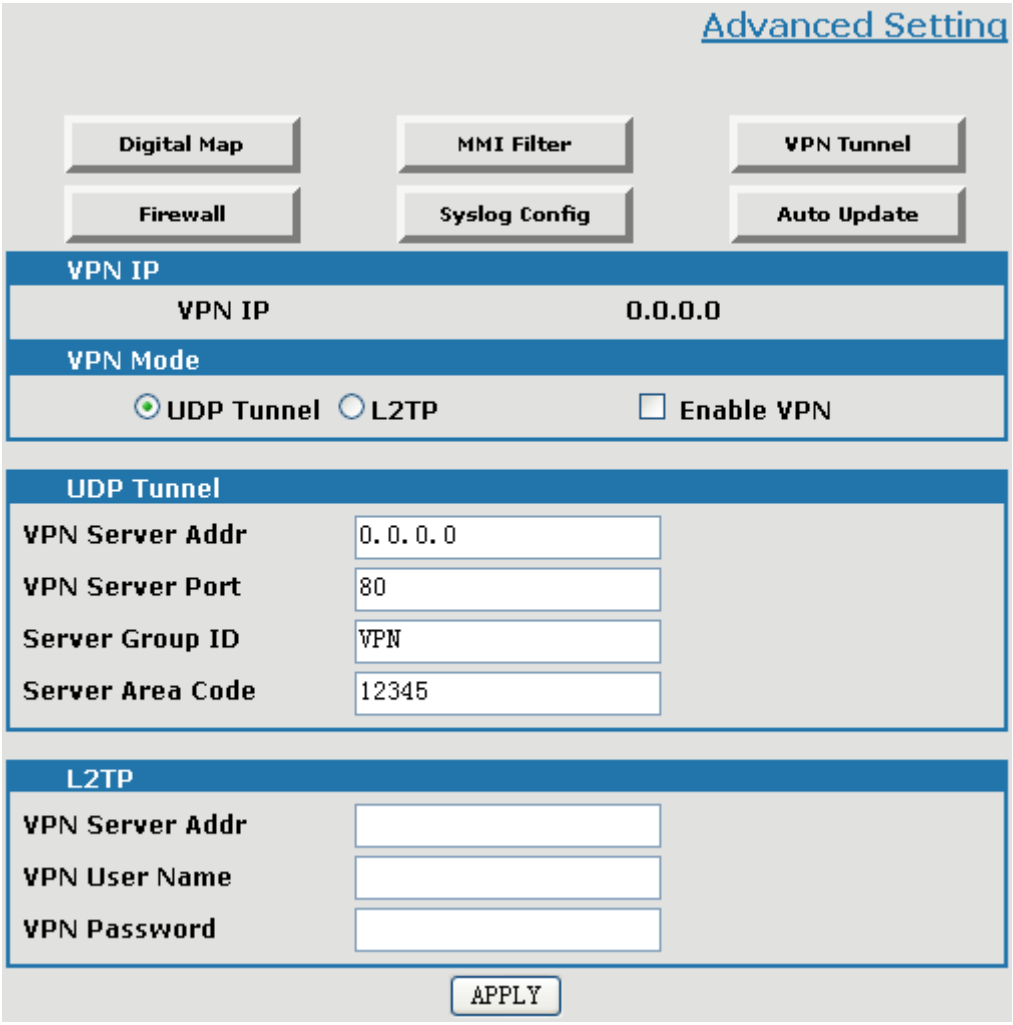
MMI filter is used to make limited access to SVG401S ATA.

When MMI filter is enable. Only IP address within the start IP and end IP can access SVG401S.



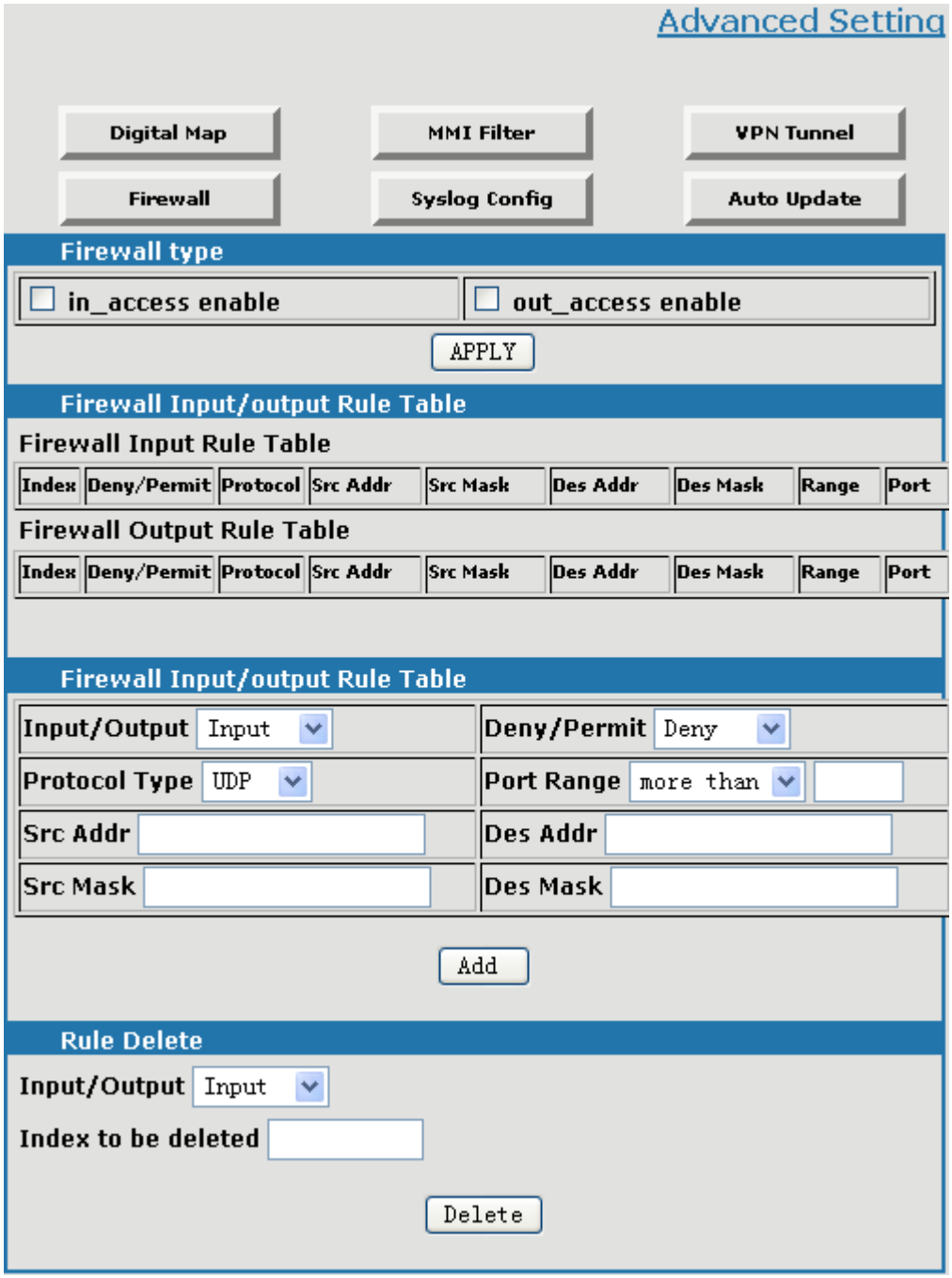
WARNING! If you try to Enable MMI filter check box without setting any item in filter table. You could not access the SVG401S ATA any more after you close this page. Therefore make sure that your IP address is included in the filter table before you enable MMI Filter.

4.9.3 VPN Tunnel configuration



4.9.4 Firewall configuration

User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.



Access list support two type limits: input_access or output_access. Each type supports 10 items maximum.

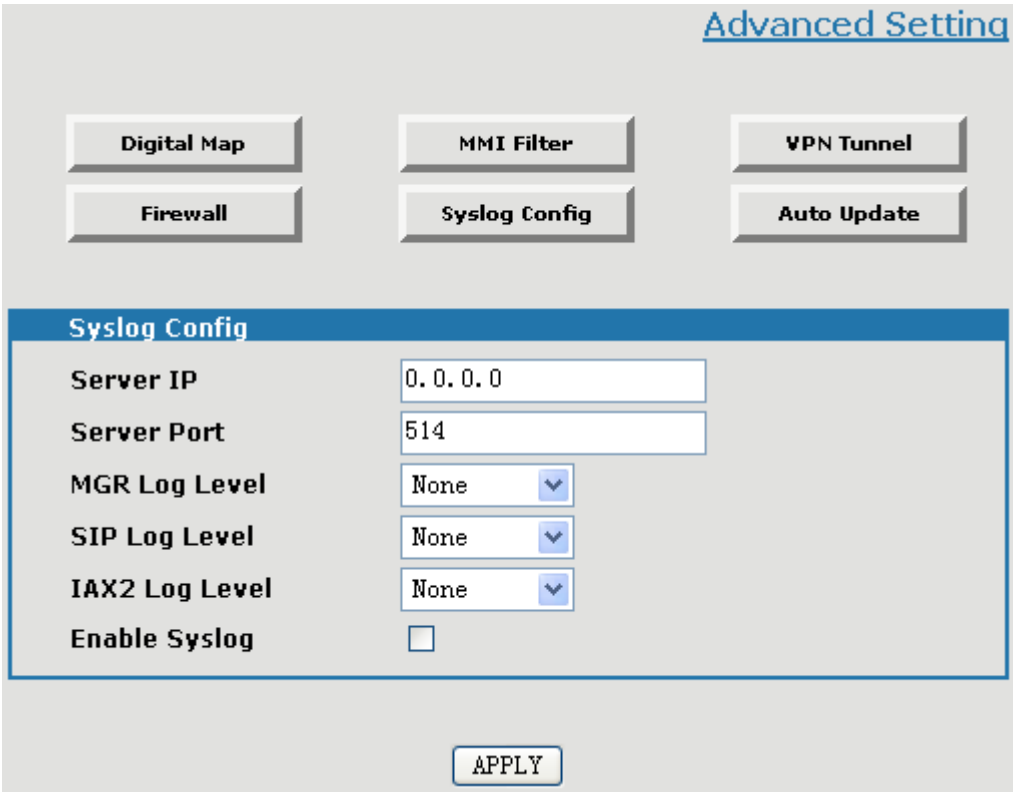
SVG401S firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:
in_access enable Enable in_access rule.
out_access enable Enable out_access rule.
Input/Output:Specify current adding rule is input rule or output rule.
Deny/Permit: Specify current adding rule is deny rule or permit rule.
Protocol Type: protocol using in this rule: TCP/IP/ICMP/UDP.
Port Range: port range if this rule.
Src Addr: Source address. can be single IP address or network address.
Dest Addr: Destination address. can be IP address or network address.
Src Mask: Source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.
Des Mask: Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network

ID.

4.9.4 Syslog configuration

Users can enable or disable syslog function and configure syslog server IP address & port.

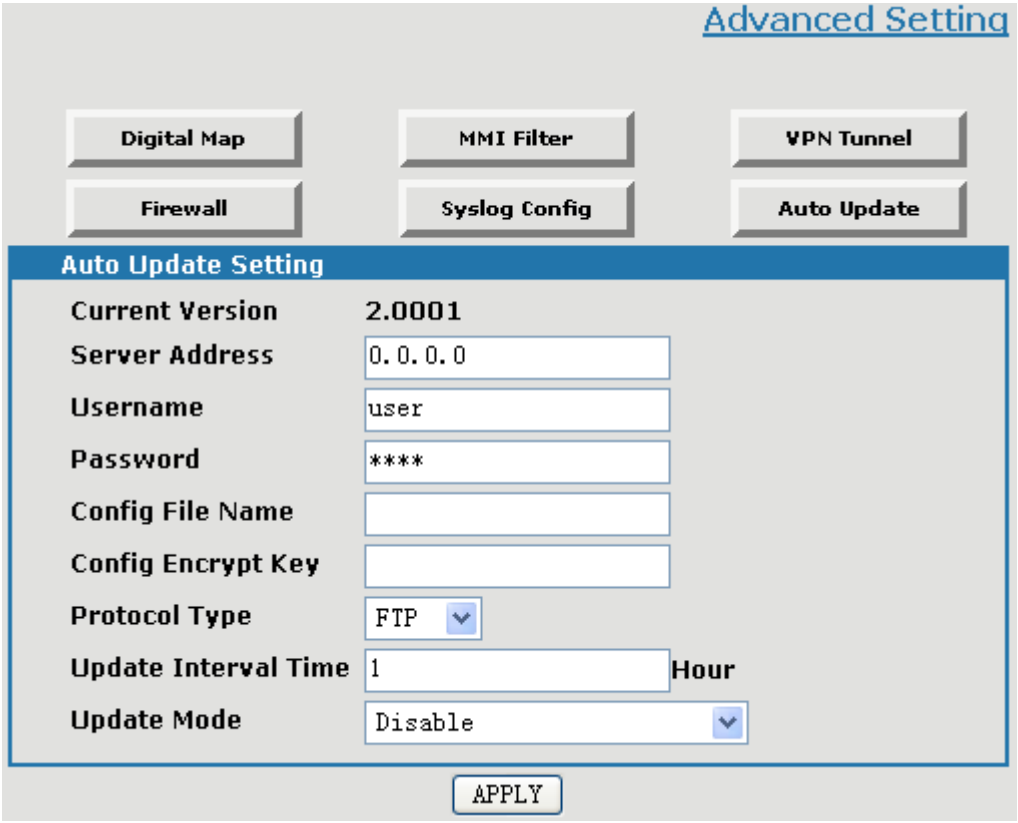


Definition of each parameter described as below

Enable Syslog	**Config enable/disable syslog function, choose it and then click "Apply" button to activate.**
Server IP	Config syslog server IP address
Server Port	Config syslog server port, click " Apply" button after inputting server IP & server port to take effect
MGR Log Level	Set system manager debug messages open with different level
SIP Log Level	Set SIP debug messages open with different level
IAX2 Log Level	Set IAX2 debug messages open with different level

4.9.5 Auto Update configuration

Inition of each parameter described as below



Server Addres: your FTP or TFTP server address.

Username: FTP server login user name. If it is used with TFTP, user need not fill anything.

Password: FTP server login user password. If it is used with TFTP, user do not need to fill anything.

Config File name: the name of config file in FTP or TFTP server. If “config file name” is unfilled, that means the phone will auto search the right config file from FTP /TFTP server by MAC address matching.

Config Encrypt Key: If the config file is encrypted, the encrypt key needs to be filled in the SVG401S ATA. config file supports AES (The Advanced Encryption Standard) 64. After download config file to phone, the phone will auto make decryption.

Protocol Type: update protocol selecting, Support FTP and TFTP protocol. Optionally, It can support HTTP.

Update interval time: the interval time of deciding to update the config file. It cooperate with update mode “update at a time interval”.

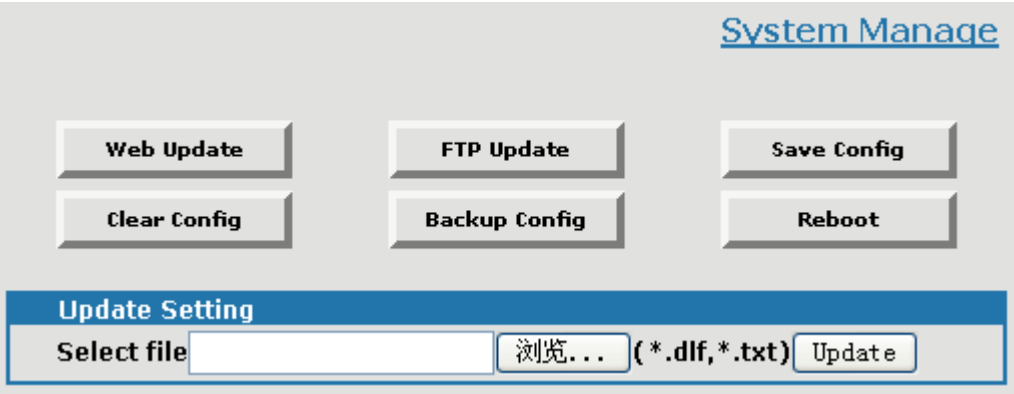
Update mode: there are three options for selection. Disable-disable auto update function; Update after reboot-when rebooting, it will auto connect server to update the config file; Update at a time interval-SVG401S will check the config file of FTP server every interval time after reboot.

4.10 System Manage

System manage is used for firmware/config file update, config file backup, save/clear config, and reboot function.

4.10.1 Web Update

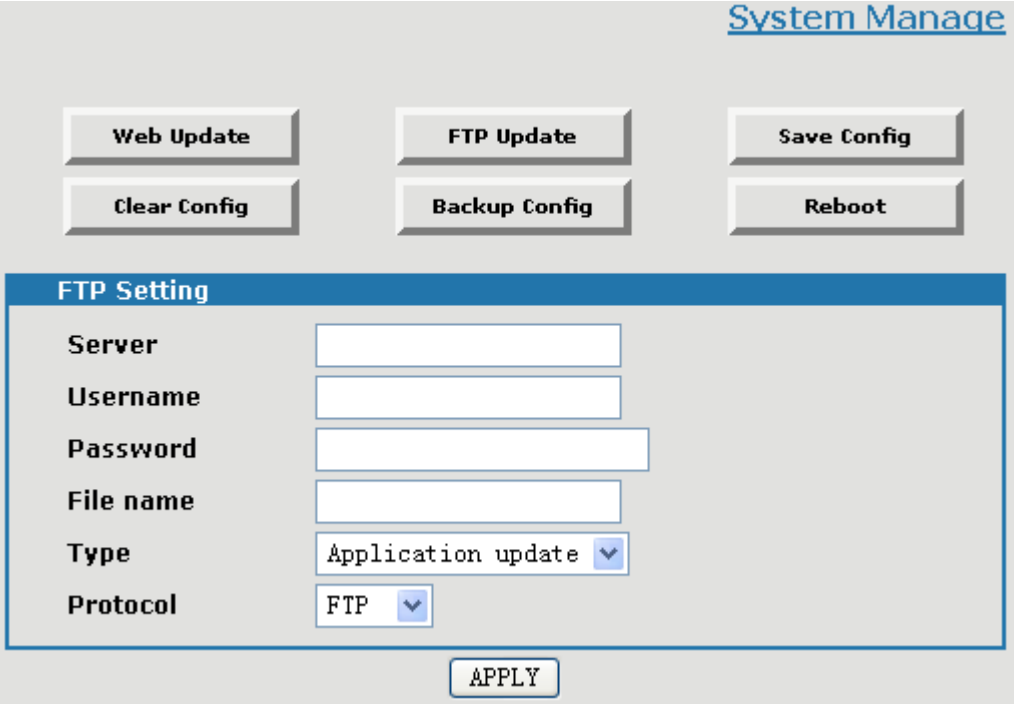
User could download the firmware and config file from pc to SVG401S.



First, click “browse” button to open window to change to the directory to the file path, then click update to download file.

4.10.2 FTP Update

If the firmware/config file is not on the local pc, but on the remoter server, user also could update by FTP/TFTP server.



Server: the FTP/TFTP server’s IP address or domain; if the file is placed under the sub directory, it could be set as this mode: 192.168.1.1/firmware/ or www.company.com/config

Username: login user name of FTP server. User would not set if server uses TFTP

Username: login user password of FTP server. User would not set if server uses TFTP

File name: firmware/config file’s name

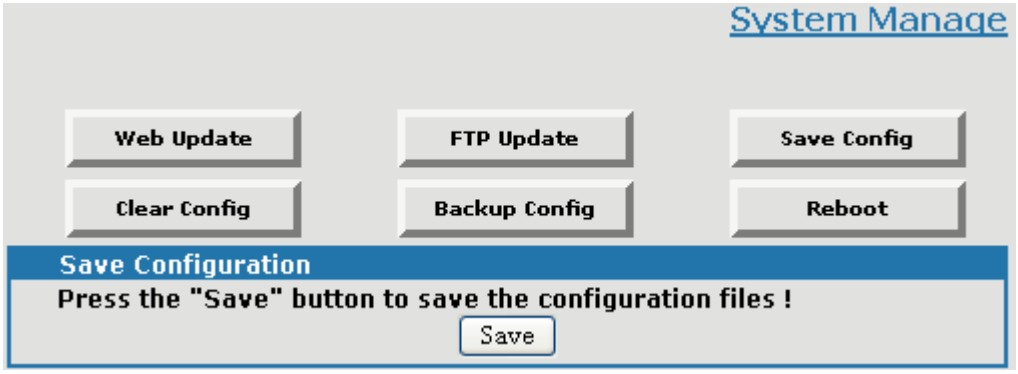
Type: what action will SVG401S execute; application update-download firmware from server; config file export-upload local SVG401S config file to server; config file import-download config file from server.

Protocol: protocol type of server

After setting, user click apply button to finish operation.

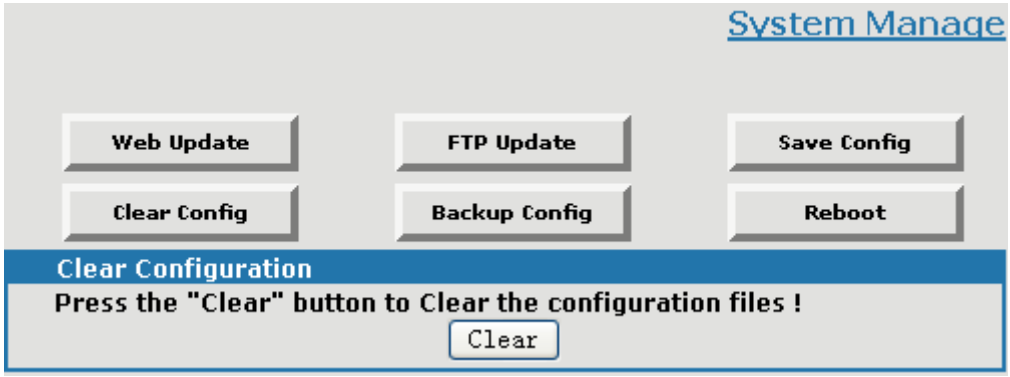
4.10.3 Save Config

After setting parameters to the SVG401S Sip ATA, user click save button to save the new configuration.



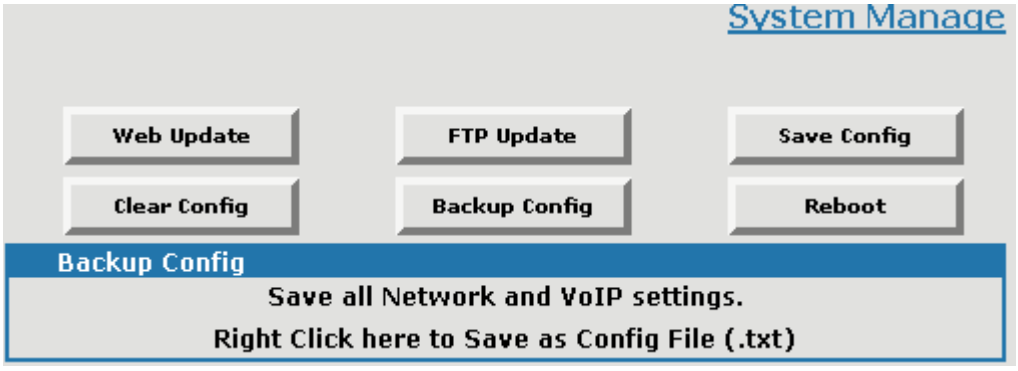
4.10.4 Clear Config

If user want to make the SVG401S return to the factory default configuration, user could login as root level(such as admin), then click Clear button to use all default set. SVG401S will clear config file in the system and reboot with default setting. If user use general level(such as guest), SVG401S will not clear all the settings, but a part of them will be cleared.



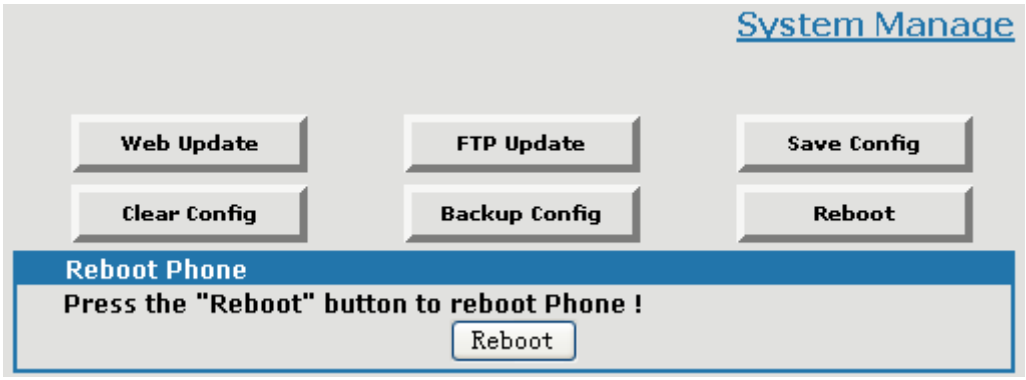
4.10.4 Backup Config

User could click the right key of mouse to save the config file of SVG401S to the local PC when mouse move to “Right Click here to Save as Config File(.txt)”.



4.10.5 Reboot

Click the Reboot button, user will make SVG401S reboot the system.



4.11 System Manage

After setting the SVG401S, user had better click Logout to end the management, keep SVG401S in safe and authorized environment.

5 Default Factory Setting

- Get IP address thru DHCP mode; WAN Port's static IP is 192.168.1.179, LAN port's IP is 192.168.10.1. Default setting is to start the DHCP service and NAT function.
- Default communication protocol is to use SIP; SIP port is 5060.
- Default HTTP port is 80; telnet port is 23.
- Default digital map rule is ending with "#" key.
- Default user accounts are admin and guest.

6 Configuration by phone

Users can configure the SVG401S SIP ATA by an analog phone:

- Input #**** Reboot Device
- Input #*000 Clear configuration
- Input #*100 ATA work at Static mode
- Input #*101 ATA work at DHCP mode
- Input #*102 ATA work at PPPoE mode

- Input #*103 ATA work at Bridge mode
- Input #*104 ATA work at Router mode
- Input #*111 Get the IP address of ATA by voice message

7 Telnet configuration

7.1 Config Procedure

- Input command “cmd” on Run submenu under PC START menu, and then input “telnet phone-IP-address, for example, telnet 192.168.10.1
- Input username and password after see the “login:”
- username and password of Administrator account are “admin”
- Configure IP Phone through command line

> ***Note:***
> 1. ***We suggest users to configure IP Phone thru web browser instead of keypad or telnet.***
> 2. ***After any change of configuration, please remember to enter “save” command to save changes and then input “reload” command to reboot IP phone to take effect.***

7.2 Telnet basic Introduction

7.2.1 Basic structure

User may use telnet commands to configure and manage the ATA:
The SVG401S ATA adopts tree structures for telnet. Every node contains its sub-nodes or local command.
User can type “help” or “?” whenever to see sub-nodes and all local command under current node.
Besides local command, there are some global commands can be used in each node.

7.2.2 Basic command

logout: exit telnet mode.
save: save current settings.
Type sub-nodes name in current node to switch to sub-node.
Type “!” or “exit” in current node to return to parent-node.
Type “help” or “?” can see all sub-nodes and all local command under current node, every help item has comments such as <command> or <node> to distinguish sub-nodes and local command. Type “help” or “?” in command can see all parameters using in this command.

When typing node name or command, user no need to key the full name, just pressing TAB button will make it more efficient.
There are two types in command parameters: optional and required. “required” parameter use “-” as prefix and “optional” use “_” as prefix. User may type “-” or “_” then press TAB button for complementarily.

7.3 Global Command

Global command is available under all nodes, SVG401S support following commands.

Command Name	Command function	Example
chinese	Set the language of help prompt info to Chinese	#chinese
clear	Clear screen	#clear
english	Set the language of help prompt info to English	#english
exit	Go back to upper level of node	#exit
help	1.Display help prompt info 2.Display all subnode under current node and local command	1. #help ping 2. #help
history	Display the history info for inputting command	#history
logout	Exit telnet config interface	#logout
ping	One test program using to check network or program availability	#ping 192.168.10.1
tree	Print out the tree structure of current node	#tree
who	Display current users login to PC	#who
Write	Save configuration to flash	#write
Reload	Reboot IP phone	#reload

Common Network command

Command Name	Command function	Example
ping	The same with above stated	#ping www.stephen-tele.com
tracert	Print out the network path	#tracert www.stephen-tele.com
show basic	Print current configuration status table	#show basic
show ip route	Print phone Router table	#show ip route
show ip arp	Print phone arp table	#show ip arp
show ip netstat	Run Netstat program	#show ip netstat
telnet	telnet another user host	#telnet 192.168.1.2
Set default	Clean phone configuration modification and restore default except network configuration	#setdefault
Set all default	Clean all configuration and restore default manufacturing setting	#setdefault all

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